

2. The Telecommunications Network

2.1. Customer Premises Equipment

2.1.1. Introduction

The key aim of telecommunications networks (voice and data) is to provide services to the user, normally referred to as the subscriber. Subscribers access these services from end-user terminals, which are mostly purchased by the subscriber from the service provider or from the third party vendor that specializes in such equipment. The end-user terminals allow the user to have access to the offered services. Service providers may provide such end-terminals, for a reduced fee or through a contract on monthly repayment terms, in which case the service provider actually owns such devices. Examples of end-user terminals include telephone sets (analog and IP), computers, workstations, laptops, switches, routers, gateways, modems (DSL, cable TV, PLC, etc), mobile handsets, fax machines, etc. All these devices are termed "*Customer Premises/Provided Equipment*" (CPE), which refers to any terminal plus its associated equipment and inside wiring located at a subscriber's premises, and connected with the service provider's telecommunications network at the demarcation point. The demarcation point simply separates customer premises equipment from the provider's equipment. This demarcation point can be either in the building or outside in the neighborhood, depending on the architecture preferred by the service provider. Due to the increased importance of telecommunications in our society, customer premises equipment, especially end-user terminals, have to also cater for people with disabilities, i.e. there must be specialized CPE (SCPE) [1] for such individuals. Different countries adopt different procedures in managing the CPE, e.g. in the US, CPE is owned and operated by the user and is beyond the direct control of the network service provider; whilst in other countries, the CPE can be provided and owned by the service provider.

2.1.2. Types of Customer Premises Equipment

Customer premises equipment can be classified according to the functions that they perform and services they offer to users or subscribers. These services or functions are of two types namely:

- a) Access functions – these devices offer the subscriber access to the service provider's network. Most of them are located in the customer's premises and are mostly purchased by the user. Though they offer access capabilities, they are not normally classified under access networks, but under CPE, due to their locations and ownerships. Examples include [2, 4]: gateways, modems (PLC, xDSL, cable TV, etc), routers, PABX, etc.

- They act as demarcation points between the provider's network and the customer's equipment/terminals.
- b) End-User Terminal functions – these devices enable the subscriber to use the actual provided services. The user interacts directly with the terminals. They are purchased by the customer and are the sole responsibility of the subscriber, and are located within the houses of the subscribers. Examples include: telephone sets (analog and IP), mobile handsets, PDAs, PCs, fax machines, TV sets, radios, etc. Figure 2.1 and 2.2, show these two types of customer premises equipment.

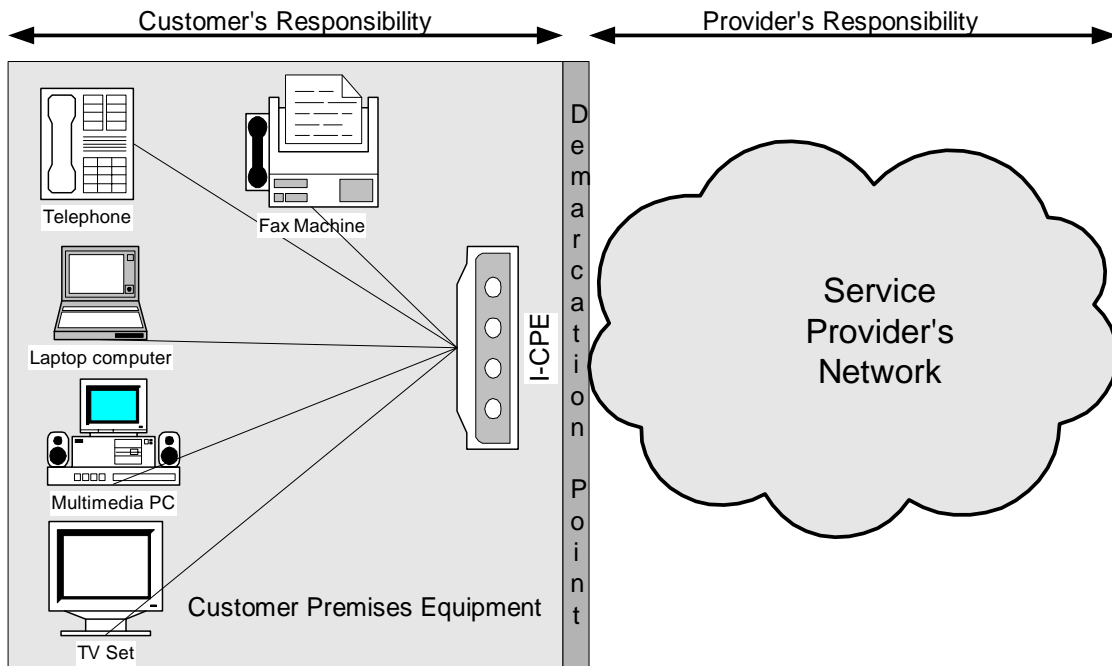


Figure 2.1, shows the functions of CPE and the responsibilities of both the customer and the service provider

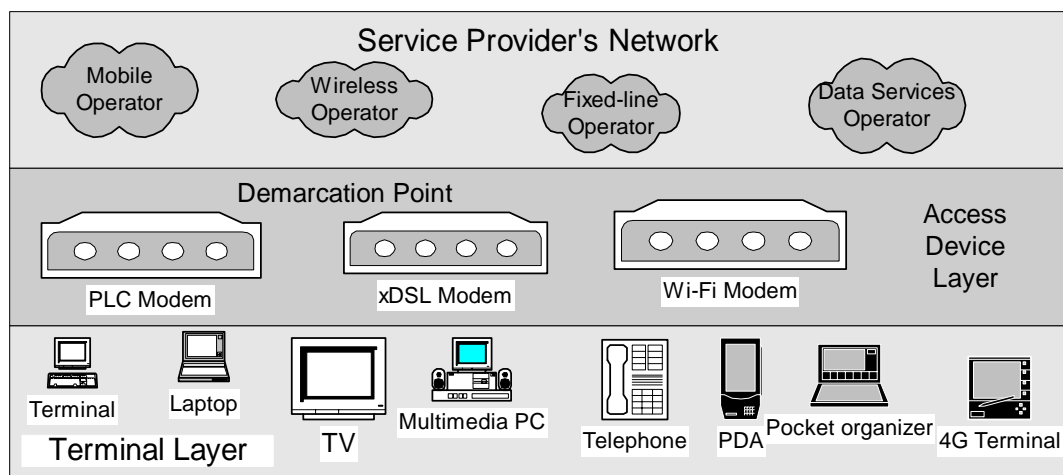


Figure 2.2, shows the layered functions of CPE, demarcation and service provider's network

Depending on countries, service provider's responsibility might extend beyond the demarcation point and cover also the access devices (as done in the US), or may remain as in figure 2.1 (as done in most other countries). In pure telephony (traditional voice), the exchange, which connects several subscribers in a certain region to the telephone company's network, is the responsibility of the service provider, hence as access device on the access network and not in the customer's premises. In Lesotho, LTA [3] recognizes WLAN equipment as part of the customer premises equipment and may be sold by interested providers.

2.1.3. Evolution of Customer Premises Equipment

Traditionally networks were built to offer a single service, the so called "*vertical integration*" concept. Each service provider had to offer a single service or was able to only offer a single service, or their networks were capable of offering only a single service. There was infrastructure for voice networks and a separate infrastructure for data networks, each with their own personnel and management. Even today, most organizations and businesses are still following this same arrangement. With vertical integration, traditional customer premises equipment were not integrated, i.e. each was only capable of supporting a single service. Traditional end-user terminals were also very limited in capability e.g. analog telephone sets were just for basic voice communications, facilities like voice message recording, user screens, call diverts, call waiting, multi-party callings, were not supported. But today, most telephone have a small screen where a caller actually sees the number being dialed and have other added capabilities in order to enhance the user experience. Television sets were small in screen sizes, black and white in color, with very poor resolution and picture quality. But they have now evolved into flat color screens with high picture quality and resolutions, e.g. high definition television (HDTV). The convergence of services has also helped into the integration of customer premises equipment, especially those responsible for access functions, such as modems. As an example PLC, xDSL, and cable TV modems support voice, data and video (triple-play capable). In addition to these access devices, end-user terminals, like IP telephones, Wi-Fi telephones are very advanced and have much more functionality. Mobile handsets for the analog 1G were very huge in size, yet very limited in functionality. The improvement started when a digital 2G was adopted and 2G capable handsets were developed. From there, 3G handsets with cameras and capable of MMS are now common, thus showing the evolution of GSM to being capable of triple-play. Traditional computers were used for data communications, but now, they act as multifunctional end-user terminals capable of being used for any customer need from data to telephony to video. Due to the multi-functional nature of computers, most end-user terminal manufacturers are attempting to develop terminals with the same capabilities as the computer. Several examples include the current mobile handsets, which now come with Microsoft operating systems, some sort of a keyboard and

functionalities mimicking the actual computer. The key trend in the CPE field is to provide integrated services, such as multi-media, voice and data services.

The following quotation from [2], summarizes the whole traditional view of CPE.

“Over the decades, due to technology limitations and regulatory restrictions, information services have been delivered to subscribers through multiple service providers via twisted-pair copper, coax cable, terrestrial wireless, satellite, and fiber optics. As a result, houses are filled with multiple Customer Premises Equipment (CPE) such as POTS (Plain Old Telephone System) phones, cordless phones, fax machines, answering machines, set-top-box, Caller ID receivers, and Direct Broadcast Satellite (DBS) receivers, etc., and the sophisticated wiring necessary to connect them. Not only is it deemed too troublesome for subscribers to use and manage all these devices, but it is also not cost-effective, as network resources are not efficiently used”. [2]

2.1.3.1. Changes in Telecommunications

Several changes in the telecommunications industry have led to the evolution of the customer premises equipment, especially the access devices. These changes include the following:

- a) Digital communications, which enabled service integration;
- b) Deregulation of the market and abolishment of monopolies;
- c) Emergence of the Internet as the medium to carry voice, video and data, thus enabling true convergence;
- d) Adoption of broadband services and their demand by customers.

Customer premises equipment are now expected to be broadband capable, thus supporting triple-play, while at the same time leaving basic services intact and should provide subscribers with high-speed connections to the Internet. The ability to support triple-play has seen the adoption of broadband-capable access devices, referred to as, “Integrated CPE (I-CPE), Integrated Access Device (IAD), and Residential Gateways (RG)” [2]. A residential gateway is mostly used when referring to CPE in the residential areas. I-CPE has to be capable of supporting all traffic types from data, voice and video; and examples include: PLC modems, xDSL modems, Cable TV modems, etc. I-CPE should be future proof in the sense that the next generation networks (NGN), which are horizontally integrated, multifunctional and single integrated networks capable of triple-play, have to be supported. The use of I-CPE as opposed to different CPE for each service provider has several benefits including the following [2]:

- a) Lower TCO on the side of the subscriber;

- b) Reduced network management complexities;
- c) Improved efficiency of network resources;
- d) Reduced space requirements on the customer's premises.

2.1.3.2. Fourth Generation (4G) Terminals

"Cellular communications have evolved from the 1G concept all the way to the 3G concept, but there seems to be no stop as of yet with the concepts of 4G and 5G already being talked about. The 4G concept is expected to evolve from several paths with 3G being the path advocated by the cellular centric industries and WiMAX, the other path. WiMAX and W-Fi offer the fixed broadband wireless technologies while 3G and 4G offer broadband wireless technologies. From these wireless technologies, it is expected that 4G will integrate all these technologies and provide a seamless access between fixed and mobile broadband wireless technologies." [5]

Every major evolution in telecommunications (especially mobile communications) comes with the new improved end-user terminals capable of supporting the new features incorporated into the technology. With this in mind, 4G is no exception and its terminals have several features including [5]:

- a) Multi-band - they have to support different bandwidth frequencies;
- b) Multi-functional - support various applications with a single terminal;
- c) Intelligent - dynamically improve its processing capability;
- d) ABC-enabled - Always Best Connected (ABC) defines the ability for a mobile user to seamlessly roam between heterogeneous wireless access networks while still maintaining connection to services (service continuity).
- e) SDR-enabled - Software Development Radio (SDR), defines software technologies that enable terminals and equipment to be reconfigured through software upgrades without replacing hardware. SDR-enabled terminals are able to scan available networks. In addition SDR will enable terminal adaptability and reconfigurability. SDR is expected to support roaming across disparate network technologies (global roaming access) in 4G systems, e.g. from Wi-Fi hotspot coverage to WiMAX or to IEEE802.20.
- f) Terminals are expected to be location and context aware.

2.1.4. Private Branch Exchange (PBX)

A PBX [4] is a circuit-switching system that has several interfaces for voice calls and data links. It was previously meant for only voice links but the latest models have interfaces for data and IP traffic, hence the name IP-PBX, which are given IP addresses just like any other device in the data network. The telephone lines connecting to the PBX are called extensions. Since PBX systems support data, they have become important to data network administrators and managers. All traffic from the different interfaces of figure 2.3, are aggregated and sent to the

telephone company's central office (CO), via multiplexed dedicated lines, which support several channels. With PBX systems, a single number is used when a user needs to make an outgoing call, as an example, at NUL we dial "9", which then connects one to the switchboard where human operators are at hand to assist in placing the call. Calls from outside (incoming calls) are automatically directed to the appropriate extensions. The caller may dial this extension directly (called direct inward dialing - DID), select it in response to an automated message, or have an attendant connect the call to an extension [4]. A PBX will also connect any internal extension with any other internal extension (station-to-station calling), at NUL a four number extension is used.

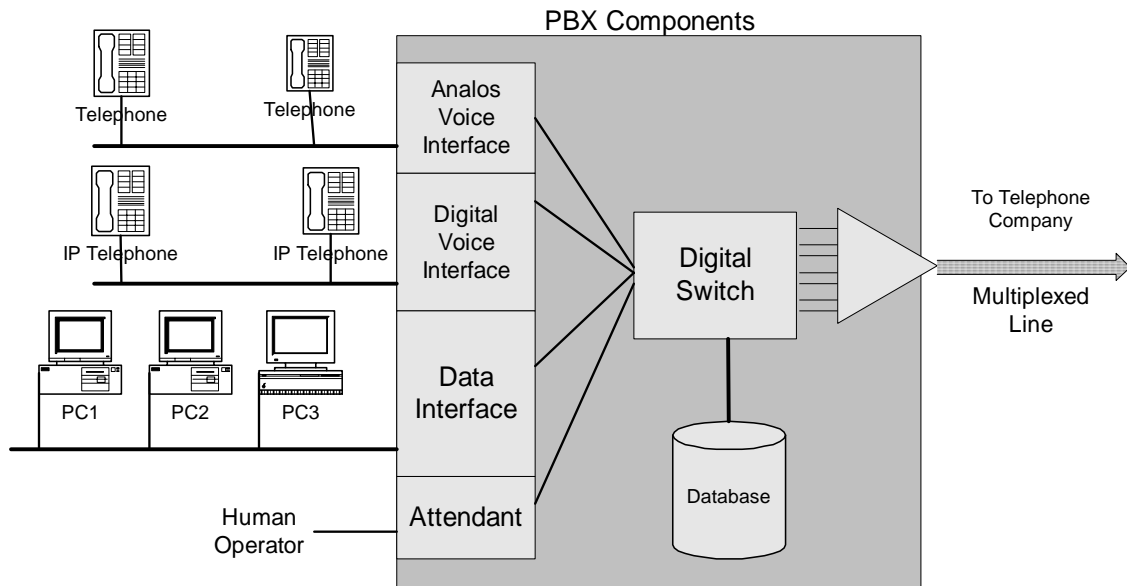


figure 2.3 [4], shows the components of a PBX system

2.2. The Access Network

2.2.1. Introduction

The evolution of customer premises equipment (end-user terminals and Access devices) is influenced by the advancement and innovations in technologies adopted in the access network. The access network simply provides the link between the customer premises equipment and the service provider's transport or core network. In other cases, the service provider may be using the third-party transport network, as in value-added services, so it's fitting to use the carrier's transport network. The access network relies on the edge devices, such as modems to link end-user terminals, but in this course, the modems are viewed as CPE based on their location in the customer's premises and responsibility by the customer. Access networks have also evolved over the years from the narrowband (low speed), single-carrier, vertical access networks to the current and future broadband (high-speed), multifunctional, horizontally-integrated access networks. The traditional access networks only carried a single service, hence each required its own access CPE devices. However, the current horizontally integrated access networks support data, voice and video (triple-play) and require a single integrated access device. The downfalls of the traditional CPE have already been discussed and they apply to the access networks as well. Traditionally the access network, also called the first/last mile, was dominated by copper cables, but now fiber cables and wireless media are taking over. This evolution is driven by the need to support multimedia communications, triple-play and the future unknown applications, which are expected to require even higher bandwidths and other network resources.

According to [6], access network is defined as; *"a network entity providing access capabilities for various service applications (a wide variety of types of user terminals) to access various service providers (specific service nodes) located at the edges of the core networks"*. Several technologies are used in the access network including: GSM (1G, 2G, 3G, 4G), Wi-Fi, WiMAX, xDSL, PLC, Cable TV, etc.

2.2.2. Functions of the Access Network

Several functions of the access network have been identified and can be grouped under five (5) major functions as follows [6]:

- a) *User Port Function (UPF)* – adapts the specific user to network interface (UNI) requirements to the core and access management functions. Examples of user port functions include:
 - Termination of the UNI functions
 - Analog to digital conversions
 - Signaling format conversions
 - Activation/Deactivation of the UNI

- Handling UNI bearer channels
 - UNI testing.
- b) *Service Port Function (SPF)* – adapts the requirements defined for a specific service-node interface (SNI) to the common bearers for handling in the core function and selects the relevant information for treatment in the access network system management function. Examples of service port functions include:
- Termination of the SNI functions
 - Mapping of the bearer requirements and time-critical management and operational requirements into the core function
 - Mapping of protocols if required for a particular SNI
 - SNI testing.
- c) *Core Function (CF)* – adapts the individual user port bearer or service port bearer requirements to the common transport bearers. Examples of the core functions include:
- Access bearer handling
 - Bearer channel concentration
 - Signaling information multiplexing
 - Circuit emulation for the ATM transport bearer.
- d) *Transport Function (TF)* – provides for the transport of common bearers between different locations in the access network and media adaptation for the relevant transmission media used. Examples of the transport functions include:
- Multiplexing
 - Cross-connection
 - Physical media functions.
- e) *Access Network System Management Function (AN-SMF)* – coordinates the provisioning, operation and maintenance of the UPF, SPF, CF and TF within the access network.

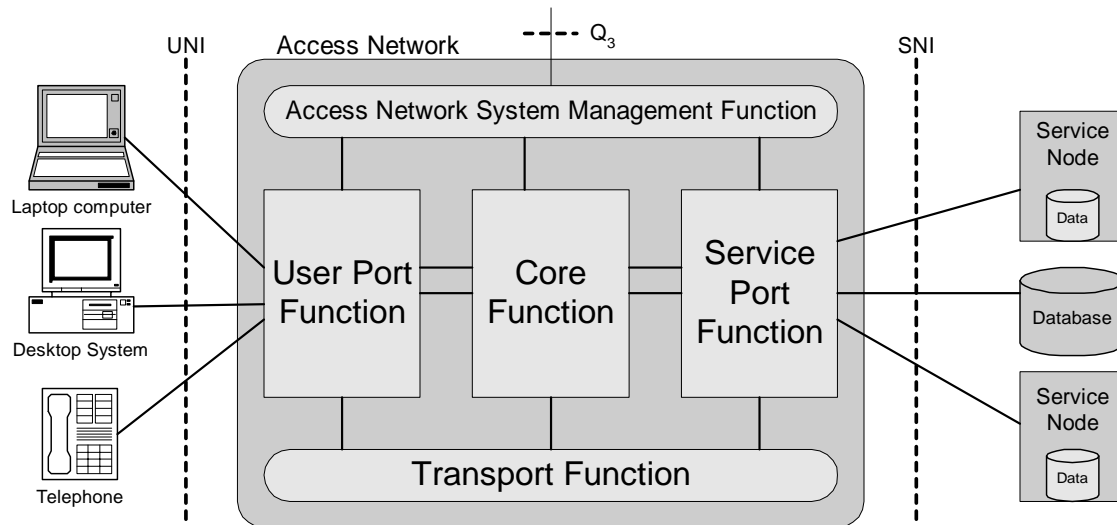


Figure 2.4 [6], shows the Access Network Architecture based on five key functions

2.2.3. Boundaries of the Access Network

The access network is part of the main telecommunications network and interfaces with all other parts at three (3) different boundaries as follows [6]:

- a) UNI - separates the CPE end-user terminals from the rest of the access network.
- b) SNI or SPI [7] - links the UNI to the actual service nodes. SNI/SPI is an open interface for use by different service providers.
- c) Q_3 - provides the interface to the telecommunications management network, for management activities in the access network.

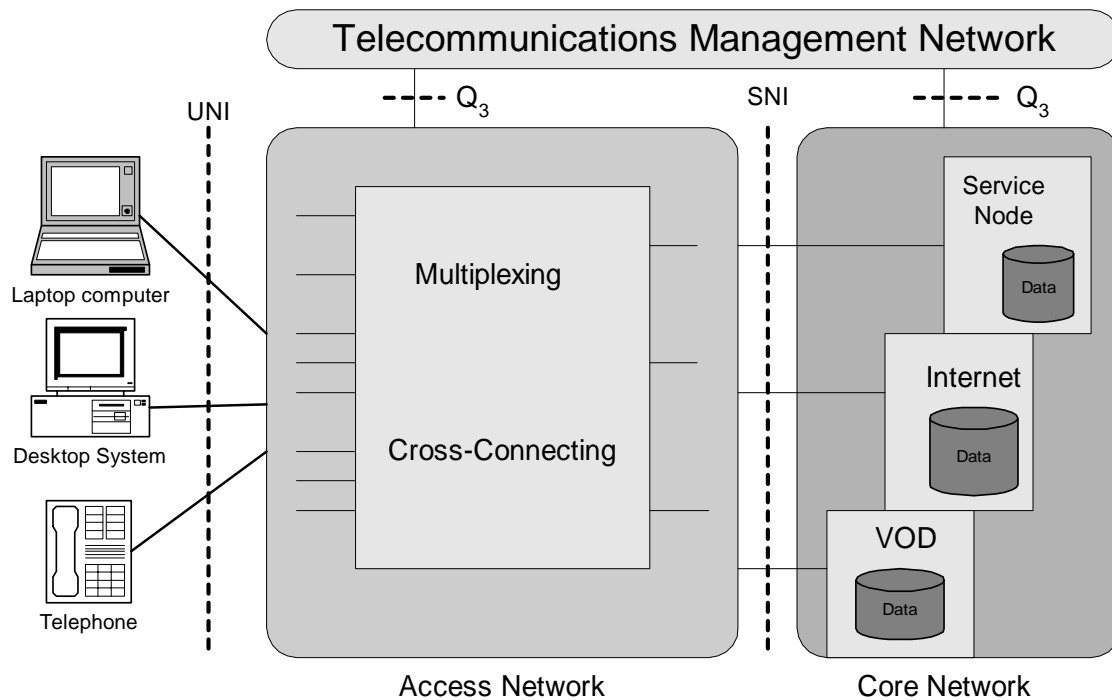


Figure 2.5, [6], shows the Access Network Boundaries

2.2.4. Access Network Evolution Drivers

The access network is evolving due to the need for new services by subscribers, which are more bandwidth hungry than ever before. Several other motivations for evolution in the access network include [7, 8]:

- a) New markets and services
- b) Technological innovation, advancements and developments
- c) Core/transport network evolution
- d) Changing regulatory environment – deregulation and unbundling of the subscriber loop
- e) Political initiatives, universal service obligations – ITU-D's Global connect: connecting the unconnected by 2015, is one of such initiatives
- f) Promoting a modern business infrastructure
- g) Need for a managed network
- h) Just-in-time network provisioning, scalability of solutions
- i) Convergence of data and voice networks – NGN initiatives
- j) Reduction of costs in the subscriber loop and management overheads
- k) The need to support triple-play
- l) Increased competition in the highly competitive and deregulated market
- m) Exploding growth and world-wide use of the Internet and its rich content.

2.2.5. Access Network Evolution Phases

“As the demand for higher-speed access to the telecommunications infrastructure grows, service providers must enhance the capabilities of the access network. Customers towards the year 2000 will no longer be willing to be limited to the 3kHz bandwidth of analog voice frequency circuits or even by the 2B+D of basic rate ISDN. Internet access, LAN interconnect for business users, video and multi-media communications all require a large pipe to the end user customer” [8].

The evolution of the access network can be studied based on the phases through which it has passed. These phases are classified based on: the media used, the topology of the network, network management capability, protection and recovery mechanisms and services carried. Based on [7], the following phases are discussed:

- a) All analog network – with this phase, the copper cables [7, 8] are used for transmission of voice (telephony). The topology followed is a star [7, 8]; with a pair of cables radiating to each subscriber. There are no inherent protection and recovery mechanisms. The management used is reactive. Since it was meant for voice, digital data services are offered by the use of modems, as in dial-up services at 28/56kbps.
- b) Narrowband digital access – in this phase, digital transmission in the form of narrowband ISDN via primary rate interfaces and leased line facilities at 64kbps. Mostly used for PBX connections to the end exchanges.
- c) Broadband Access network – in this phase, the capacity of copper has been increased to support technologies such as xDSL (ADSL, VDSL, etc), now with speeds up to 60Mbps. In the broadband era, several other technologies are used including wireless, PLC, Cable TV, fiber technologies, etc.

2.2.6. Access Network Alternatives

“Designing a cost-effective customer access network represents a serious engineering challenge. However, a large number of alternative technologies and architectures are available” [8].

There are several factors to consider before deciding on which access technology to deploy, and these factors include among others [8]:

- a) Economic considerations,
- b) Services to be offered,
- c) Competitive environment,
- d) Availability of the basic technology for the local environment,

- e) Customer density, location and distribution,
- f) Imbedded equipment base
- g) Health issues
- h) Political and regulatory issues
- i) Scalability, reliability and robustness
- j) Environmental issues (global warming, community eyesores, etc).

The access technologies are grouped based on the underlying physical medium used in each technology, and are traditionally grouped as follows [7, 8, 9]:

- a) Copper cables
- b) Optical fiber cables
- c) Wireless medium, relying on the air interface
- d) Hybrid schemes, based on a combination of different media.

2.2.7. Copper Cables

Copper cables are grouped as twisted pairs and coaxial (coax) cables. Twisted pairs in turn are further divided as unshielded twisted-pair (UTP) and shielded twisted-pair (STP).

2.2.7.1. Twisted-Pair Cables

One of the most popular medium is through the twisted-pair copper cables, which are the foundation of the traditional PSTN. But even today most of the Internet and data connections in companies are through the use of local loops based on twisted pairs. LANs in buildings are now based on twisted pair cables with categories from 1 to 7, and mostly using UTP. Despite its huge base, twisted-pairs are highly susceptible to noise, interference and distortion, including electromagnetic interference (EMI), radio frequency interference (RFI) and the effects of moisture and corrosion [9]. So, when deciding to use twisted-pair cables, the engineer has to consider the age and health of the cables. Though UTP is the mostly used medium for desktop wiring, they are eventually expected to be replaced by optical fiber cables, as the future of desktop wiring. Twisted-pair cables need a lot of repeaters along the channel and this results in many components to be maintained, thus increasing long-term operational costs. The performance of twisted pair cables is inversely proportional to the distance covered, i.e. the longer the distance, the greater the impact of errors and impairments, which diminish data rates. The total useable frequency spectrum of twisted pair ranges from 1MHz for telephony to about 2.2MHz for DSL standards [9].

2.2.7.1.1. Applications of Twisted-pair cables

The primary applications of twisted-pairs are in [9]:

- a) Telephony

- b) PBXs between telephone sets and switching cabinets
- c) LANs, desktop wiring
- d) Local loops, including both analog telephone lines and broadband DSL.

Telephony and Dial-up access

Twisted-pair is used in traditional analog subscriber lines also known as the telephony channel or 4KHz channel, to provide voice communications amongst the subscribers. The traditional PSTN is one of the most successful and widely deployments of twisted-pair in the access network. But the exploding growth of the Internet and data traffic led to the use of this analog media for the provision of digital traffic, through the use of modems, the so called dial-up copper access. Dial-up services are however limited in speeds with modem technologies allowing 28, 56 and 100kbps [8] to be transmitted over voice telephony network originally meant for the transmission of ordinary speech.

Integrated Services Digital Network (ISDN)

ISDN (precisely N-ISDN) was introduced in 1983 [9], as an all digital access network technology over the public telephone networks, intended to provide end-to-end digital services (voice and data services). There are two types of N-ISDN namely [9]:

- a) Basic Rate Interface (BRI) – composed of two bearer channels at 64kbps each and one delta channel at 16kbps. B-channels are used to carry user traffic (voice, data and fax), while the D-channel is used for signaling, but can also be used to carry low-speed packet data. So, BRI offers 2B+D totaling 144kbps over a single twisted-pair with a loop length of 5.5KM; and is used in residences and small businesses requiring only a few lines.
- b) Primary Rate Interface (PRI) – composed of 23 B-channels (T-1 in North America and Japan) or 30 B-channels (E-1, rest of the world). In PRI, the D-channel is 64kbps as opposed to 16kbps in BRI. PRI is used for business systems to connect PBXs and is offered over two twisted-pairs.

N-ISDN, with its BRI, is no longer sufficient in this age of high-speed Internet access and web surfing.

Digital Subscriber Line family (xDSL)

DSL systems rely on the existing twisted-pair copper wires originally meant for telephony in the subscriber loop. The high-data rates are made possible by the use of hardware (modems) with efficient signal processing and advanced modulation techniques, while still preserving the usual voice connectivity. When compared with dial-up access, DSL offers dedicated, fixed connection to data services, and when compared with N-ISDN, offers higher data rates, with ability to support video services. Within this family, there are symmetric services (same bandwidth in both directions) and asymmetric (unequal bandwidth, with downstream usually higher). Deciding on which member of the family to choose,

has to consider: environment, prevailing conditions, types of services to be offered, distance-limitations, etc.

High-Bit-Rate DSL (HDSL)

A symmetrical service providing T-1 or E-1 capabilities, yet cheaply deployable and covers distances of up to 3.6km. HDSL is deployable over two twisted-pairs, but in homes with a single twisted-pair, HDSL 2 has been adopted and it affords 1.5 to 2Mbps over a single twisted-pair. [8, 9]

Asymmetric DSL (ADSL)

This is an asymmetric services deployable over a single twisted-pair cable. ADSL with speeds up to 4Mbps upstream is now offered by Telkom SA, while Telecom Lesotho now offers ADSL at 1Mbps. ADSL covers 5.5km from the local exchange and data rates are inversely proportional to distances between the user and the local exchange. Two standards for ADSL are:

- a) Standard ADSL – currently deployed as a majority, with downstream speeds of 7Mbps [9] and 800kbps upstream. This is sufficient for web surfing and low grade video entertainment. It is however not sufficient for digital TV or interactive services.
- b) ADSL 2 – supports up to 8Mbps downstream and 1Mbps upstream, thus supporting digital TV and interactive services. ADSL 2+ is now available and supports up to 24Mbps downstream and 1Mbps upstream.

Symmetrical/Single-Line DSL (SDSL)

Symmetrical service that covers up to 5.5km and is deployable over a single twisted-pair, with speeds in multiples of 64kbps to a total of 2Mbps in both directions. SDSL is a good solution in [9]:

- a) Businesses
- b) Residences
- c) Small Offices
- d) Remote Access into corporate facilities.

Symmetric High-Bit-Rate DSL (SHDSL)

A version of SDSL, which supports up to 5.6Mbps in both directions [9].

Rate-Adaptive DSL (RADSL)

Can be deployed as a symmetric or asymmetric service, and dynamically adapts the data rate based on any changes in line conditions and loop length. It covers 5.5km over a single twisted-pair cable and has rates varying from 600kbps to 7Mbps downstream and from 128kbps to 1Mbps upstream. [9]

Very-High-Bit-Rate DSL (VDSL)

Offers very high data rates (up to 100Mbps), but over very short distances. It is a hybrid system that relies on fiber optical cables between the central office and the head-end equipment near the customer. It is highly distance-limited as follows:

*1.5km – 13Mbps downstream
300m – 55Mbps downstream, 5Mbps upstream*

It has enough capacity to facilitate the delivery of several HDTV channels as well as Internet access and VoIP [9]. With the latest version, VDSL 2, up to 100Mbps in both directions is achievable, but over very short distances. [8,9,10]

2.2.7.1.2. Advantages and Disadvantages of Twisted-Pair

Advantages:

- a) High availability – over 1billion telephone subscriber lines already deployable [9].
- b) Low cost of installation on the premises
- c) Low cost for local moves and changes in places.

Disadvantages:

- a) Limited frequency spectrum
- b) Limited data rates
- c) Short distances required between repeaters
- d) High error rates.

2.2.7.2. Coaxial Cable (Coax)

The difference between coaxial cable and the twisted-pair is that the actual copper wire in coaxial cable is thicker than that in twisted-pair. In addition, coaxial cable is unaffected by surrounding wires, which contribute to EMI and as such provides higher transmission rates than twisted pairs. Coaxial cable has more frequency spectrum (370MHz, 750MHz or 1GHz) than twisted-pair, and can afford 370 to 1,000 times more capacity than single twisted-pair [9]. It requires amplifiers every 2.5km, though much better than twisted pair, it still results in much costs due to the substantial number of amplifiers deployed throughout the network. Mostly used by cable TV operators, it is deployed as bus architecture, resulting in the sharing of bandwidth among subscribers, which in turn leads to increased congestion levels as more users join the network (i.e. it is not scalable). High security risks, which are inherent in the bus topology.

2.2.7.2.1. Applications of Coaxial cable

Several applications are identified in [9] as follows:

- a) Telephony networks as interoffice trunks
- b) Submarine cable carrying international traffic

- c) Used in LANs between 1980 – 1987
- d) Used in cable TV and in the local loop in the form of HFC architectures.

2.2.7.2.2. Advantages and Disadvantages of Coax

Advantages:

- a) Broadband capable
- b) Greater bandwidth
- c) Greater channel capacity
- d) Lower error rates compared to twisted-pairs
- e) Greater spacing between amplifiers compared to twisted-pairs.

Disadvantages:

- a) Problems with the deployment architecture –susceptible to congestion, noise and security risks
- b) Bidirectional upgrade required – originally meant for broadcasting, so for interactive services, they have to be upgraded
- c) Great noise
- d) High installation costs
- e) Susceptible to damage from lightning strikes.

2.2.8. Optical Fiber Cables

Optical fiber operates in the visible light spectrum, in the range from 10^{14} Hz to 10^{15} Hz and cover longer distances as they require repeaters after 800km or recently up to 6,400km [9]. Fiber offers unlimited bandwidths in data transmission and is interference free as opposed to copper cables [15].

2.2.8.1. Advantages and Disadvantages of fiber

Advantages [9, 15]:

- a) Extremely high bandwidths – offers far more bandwidths than any other cable-based medium in magnitudes of 100/1000 more than copper cabling.
- b) Elastic traffic – varying capacity, with improvement in technology more capacity can be added without having to replace the original fiber, hence why it's viewed as the access medium of tomorrow and beyond; and provides the next step for high-bandwidth scalable connectivity.
- c) Virtually immune to EMI, and thus have a very low bit error rates, 10^{-13} , which means fiber-optic transmissions are virtually noise-free.
- d) Secure transmissions, due to its inherent security provided by point-to-point connections.
- e) Low in weight and mass – less human installation power than traditional copper cables.

- f) Cover longer distances (up to 6,400km) than any other cable-based access technologies.
- g) Ensures that today's networks can support tomorrow's applications, which are unknown, but expected to be even more bandwidth-hungry.
- h) Enable solutions that make real-time remote access possible.

Disadvantages:

- a) Though dropping (60% per annum), fiber installations are still high.
- b) Special test equipment required
- c) Vulnerability to physical damage - easily cut-off during construction activities
- d) Vulnerability to damage caused by wildlife, such as birds that use outer covering for their nests.

2.2.8.2. Applications of fiber

Several applications of fiber include [9, 14, 15, 17]:

- a) Public and private network backbones are fiber-based
 - Backbones of the PSTN worldwide have been upgraded to fiber
 - Backbones of Internet service providers are fiber-based
 - Cable TV systems rely on fiber as well
- b) Electric power utility companies rely on optic fiber to direct and control power distribution
 - Excess capacity of fiber is left after distribution of electricity
 - This left excess capacity is resold or leased to interested parties including telcos
 - Dark fiber, which is the leasing of only fiber pairs without the active elements, which become the responsibility of the company acquiring the fiber
 - With dark fiber, costs are for the physical media and not the bandwidth.
- c) Another application of fiber is in the local loop, with several arrangements, hybrid fiber coax (HFC), VDSL, etc.
- d) Applied in LANs, with fiber distributed data interface (FDDI), the first optical LAN backbone that offered 100Mbps capacity. Another example is the linking of NUL computer laboratories to the PBX.
- e) Used in telemedicine applications requiring extremely high resolutions for video and imagery.

- f) Now applied in home area networks (HANs) – new homes in developed countries are now wired with fiber so that distribution of rich content is possible.

2.2.8.2.1. Fiber in the local loop

Fiber is now penetrating in the local loop and there are several drivers for the adoption of fiber including [13]:

- a) Economic – dropping costs of fiber at 60% per annum [9]
- b) Social – education, with interactive video, online classes, distance education and the Internet.
- c) Entertainment – interactive video, video on demand and IPTV.
- d) Business – e-commerce, video conferencing, and triple-play.

Despite these drivers, fiber deployment is constraint by the costs associated with its purchase, active equipments and civil works (especially for underground deployment), and regulatory conditions, which differ from country to country [14]. Fiber in the local loop takes different forms in terms of architectural deployments. These different architectures are due to mainly:

- a) Costs of buying the fiber cable, cabinets, active equipment, etc
- b) Civil works involved
- c) Disruptions in subscriber homes
- d) Legal issues for land acquisitions and digging
- e) Time needed for deployment to be completed
- f) Services needed over fiber or the amount of traffic that needs to be carried.

Several architectures, which depend on the penetration of fiber, are termed “fiber to the ... (FTTx), and include [10, 12, 13, 14]:

- a) **Fiber to the home (FTTH)** – where fiber optical cables are used all the way from the central office to the subscriber’s home, i.e. fiber is terminated within the premises of the customer. From there, any other in-house wiring media or wireless can be used to connect to the subscriber’s end user terminals. FTTH supports up to 50-100Mbps, thus offering triple-play and future unknown applications. Inherent security is also offered as there is a point-to-point link from the subscriber to the central office. It is sometimes referred to as FTTP, for premises. FTTH is currently the most popular access alternative in Greenfield areas, with most new homes in developed countries, having fiber thus increasing their real estate value. FTTH for desktop is the latest extension of FTTH to the end-user terminals, such as workstations and laptops.
- b) **Fiber to the building (FTTB)** – where fiber optical cables are used all the way to the building (e.g. basements of tall buildings). After terminating fiber at the basement, services are carried to subscribers in all floors of the

building either by VDSL or Ethernet over copper. Costs of FTTB are very affordable as businesses/subscribers occupying such high rise buildings can share the costs of bringing fiber to the buildings. FTTB is very popular in high rise buildings and countries with many large apartment buildings, such as India, Korea, Northern Europe and Taiwan.

- c) **Fiber to the Curb/Neighborhood (FTTN/C)** – where the optical fiber stops at a street-side cabinet and services continue to the subscriber via VDSL or VDSL 2 over copper cables, FTTC is attractive in brownfield areas, where operators already have copper installed into the premises.

Each of these architectures is useful in the access network and the actual choice of which to implement depends on the already identified factors, plus the business model available to operators.

In further trying to reduce the costs of fiber, new architectures employ what is called passive optical network (PON), which helps to lower costs per home subscriber due to the sharing of fiber [12]. PON is shown as figure 2.6, and has the following advantages [14]:

- a) Reduce costs due to the sharing of opto-electronics
 - Optical line terminals (OLT)
 - Substantial part of the fiber cable plant
- b) Reduced cabling
- c) Reduced civil works, which form a prohibitive factor in fiber deployment.

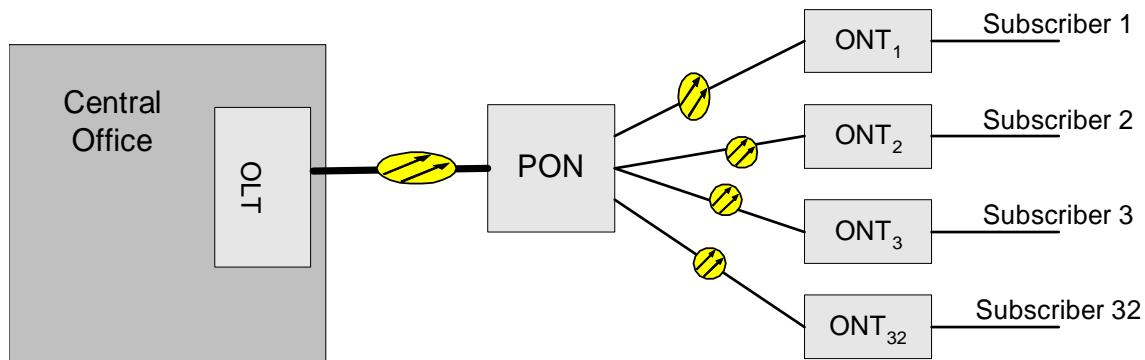


Figure 2.6, adapted from [10, 12, 14], shows a typical FTTx using PON technology

PON has now been improved to GPON for Gigabit passive optical networks. From figure 2.6, PON provides a point-to-multipoint technology in which a single optical port (OLT) communicates with multiple subscriber devices (ONT), which terminate the fiber on the subscriber terminals. In point-to-point architectures, each OLT is connected directly to a single subscriber's ONT, which is more expensive than point-to-multipoint architectures afforded by PON and GPON.

2.2.9. Wireless Media

Though section 2.2.6 classified access technologies based on either copper, optical cables hybrid technologies and wireless; they can be classified into two as: wired and wireless. In wireless medium, there is no guidance as the air interface is used to transmit signals, hence why it's referred to as unguided medium. Wired solutions have fixed connections which help in guiding the traffic, hence why they are referred to as guided or closed medium. Though they have their detractors, wireless media offer alternatives where it's not cost-effective or feasible to deploy cables, mostly in rural areas. They should not be seen as competitors, but rather viable alternative solutions, where other media cannot be deployed. Wireless media rely on radio, microwave and infrared frequencies to transmit signals through air and space [17], with each technology having its own frequency spectrum or range on which it is able to operate. Frequency spectrum is a natural, yet very scarce resource, which is managed, controlled and allocated by the telecommunications authorities (LTA in Lesotho). Within the spectrum, some frequency ranges or bands are licensed, while others are unlicensed and can be used by anybody without consulting the authorities. In most countries, frequency spectrum licenses are very expensive and prohibit the penetration of telecommunications, as an example, 3G licenses are very expensive, thus contributing to the slow penetration and unaffordability of 3G services, which seem to be obsolete even before much acceptance due to emerging 4G technologies.

2.2.9.1. Microwave

Microwave operates in the 1-100GHz [9] frequency bands to provide very high-bandwidths. Though allocated this range, microwave systems largely uses up to 50GHz range, because at 60GHz, the *oxygen layer* is encountered, which completely absorbs the microwaves. However, developments in technology have now made possible what is called "*virtual fiber*" – where microwave systems operate in the 70-95GHz ranges but over very short distances [9, 17]. Microwave transmission systems consist of two directional antennas facing each other and mounted over towers, as they require line-of-sight (LOS). When used in the transport network, relay stations are used to extent the reach of microwave systems. The distance between the antennas is affected by the curvature of the earth, thus leading to the construction of taller towers or placement of antennas over tall buildings, which in turn infuriates the communities, who view them as eyesores and cause the value of real estate to drop. Spacing is also subjected to the frequency band, for example, at 2, 4 and 6GHz bands antennas can be spaced 72km apart; and at 18, 23 and 45GHz bands antennas can be spaced 1.6 to 8km apart [9]. In addition to this spacing, microwave systems require line of sight, which is interrupted by the curvature of the earth at about 144km. the frequency spectrum for microwaves has both licensed and unlicensed bands. With

unlicensed bands, anyone can use them without applying to the authorities for licenses, and examples include: 2.4 and 5GHz bands used for WLAN.

2.2.9.1.1. Applications of Microwave

Microwave applications can be classified into traditional and the emerging wireless broadband applications. Traditional applications of microwave originated from its successful deployment during World War II in military applications and were then introduced into commercial communications for the following [9, 17]:

- a) Deployed as a replacement of coaxial cables in the PSTN in the late 1940's.
- b) Across wide open areas where it's not practical or feasible to lay cable, such as deserts, rural and mountainous, swamps, seas and large lakes.
- c) Between two buildings in metropolitan areas as a replacement of leased lines. This is very beneficial because leased lines are billed monthly and based on coverage distance, thus resulting in a lifetime cost. With microwave, there is no monthly fee, thus more savings on the company's part. This kind of scenario is common in multi-location companies, such as, health care facilities with many clinics and hospitals scattered in a certain area; multi-campus universities; banks with several branches and governments with its scattered ministries/departments.
- d) Useful in bypassing – when an organization builds a new, additional remote facility that requires communication with the main office, microwave systems are very helpful as opposed to wiring. This is because, with wiring, *“the cost and time to get permission to break ground, lay conduit, pull cable, repave and relandscape would be cost- and time-prohibitive”* [9].
- e) Interconnection of LANs to produce a virtual LAN – in this case, microwaves bridge the two different LANs and gives a combined network the appearance of one LAN.
- f) Useful in disaster recovery applications, because of its easy of deployment.
- g) Satellite to ground links.

Though, it appears like an ultimate solution in metropolitan areas, microwave growth is limited by the fact that only a limited number of people can be operating on the same frequencies in the same area. Therefore a big limitation of microwave is potential congestion in key metropolitan areas [9]. In traditional applications, microwaves were mainly used as replacement to the copper wires in the PSTN, and the new emerging applications focus on the wireless broadband and include [9]:

- a) Wireless WANs – used to support GSM data services such as 2.5G enhanced data services (GPRS), 3G (EDGE, UMTS), 3.5G (HSDPA) and 4G mobile broadband systems.

- b) Wireless MANs – supports broadband fixed wireless access (BFWA) systems, such as WiMax (IEEE 802.16), mobile-Fi (IEEE 802.20), Wi-TV (IEEE 802.22) and virtual fiber.
- c) Wireless LANs – IEEE 802.11 family of protocols operate in the microwave band, relying on the unlicensed bands of 2.4GHz and 5GHz.
- d) Wireless PANs – used to support technologies like Bluetooth (IEEE 802.15.1); wimedia (IEEE 802.15.3); ZigBee (IEEE 802.15.4) and ultra-wide band (UWB).

2.2.9.1.2. Advantages and Disadvantages of Microwaves

Advantages [9, 17]:

- a) Cost savings – less expensive than leased lines
- b) Portability and reconfiguration flexibility
- c) Substantial bandwidth
- d) Useful where cabling is impractical and prohibitively costly.

Disadvantages [9, 17]:

- a) Line of sight requirements
- b) Susceptibility to environmentally caused distortions
- c) Regulatory licensing requirements usually takes time and a lot of politics
- d) Potential environmental restrictions – community eyesores and degradation of real estate values.

2.2.9.2. Satellite

Satellite systems follow the same concept as radio link and microwave systems, with the exception that the intermediate link system is in orbit around the earth as opposed to being constructed on the ground. Satellites are launched into space and provide coverage on earth, just like base stations cover certain areas on land called cells. The area of the earth covered by the satellite beam is called a “*footprint*” [9, 16, 17], which may be very large or only cover a focused area. Satellites are grouped based on their distance from the earth’s surface and the orbit in which they operate as follows [9, 16, 17]:

- a) **Geosynchronous Orbit (GEO)** – placed 36,000km above the earth, where they receive uplink signals from earth-based transmitters (or other satellites) and downlink those signals to earth. Due to the large distances, signals suffer a delay of approximately 0.25s in both directions, thus amounting to 0.5s, which is not tolerable in time-critical applications like voice communications. In the geosynchronous orbit, satellites orbit the earth at the speed that exactly counters the pull of gravity and have an orbit time of 24hours which, because of the earth’s rotation, gives them the appearance of being stationary. Satellites in this orbit are mainly use for telecommunications services,

especially delay insensitive data or Internet traffic. Due to their large distances, they require the most power than all other all other satellite systems, but have a benefit of providing the largest footprint with only three GEOs able to cover the entire globe, however, the delay factor inhibits the use of GEOs. New developments have now seen data rates increased up to 155Mbps, which in turn results in smaller sized earth-based receiver stations. Main applications of GEO systems include: one-way broadcast, VSAT systems, point-to-multipoint links. Since one-way broadcasts are delay-insensitive, GEO systems are used to transmit International Television.

- b) **Middle Earth Orbit (MEO)** – Placed 10,000 to 15,000km above the earth. Since they are closer to the earth than GEO systems, they turn to move much faster – in about one to two hours. Thus for global coverage, more MEOs (about five times more) are required than GEOs. Despite increased numbers, power requirements and delay (0.1seconds) are reduced. Mainly applicable in regional networks, to support mobile voice and low speed data, in the range of 9.6kbps to 38kbps.
- c) **Low Earth Orbit (LEO)** – placed 640 to 1,600km above the earth and are a lot like cellular networks, except that the cells as well as the users are moving. Since they are more closer to earth, they travel even faster to keep them from falling back to earth, which in turn means there are more LEOs than other satellites (20 times more than GEOs and 5 times more than MEOS). They have low power requirements thus supporting smaller handheld devices. Because of their rapid movement, LEOs support handover or switching of calls – i.e. a call is switched on to the oncoming satellite as soon as one is going out of coverage. They have a delay of about 0.05seconds and are used to support mobile voice, low- and high-speed data. LEOs are categorized based on their applications as follows:
 - Little LEOs – offer 2.4 to 300kbps in the 800MHz range. Useful for delivering messaging, paging and vehicle location services.
 - Big LEOs – offer 2.4 to 9.6kbps in the 2GHz range. Used to provide voice services to areas that currently have no form of network connection (terrestrial or cellular).
 - Broadband LEOs – offer 16kbps to 155Mbps in the 20 to 30GHz and support data and multimedia traffic.

2.2.9.2.1. Frequency Allocation for Satellite

Satellites use the frequency spectrum intended for microwaves, and their bandwidth is affected by the spectrum allocated, the portion which is actually used to operate the satellite and the number of transponders. The transponder is the satellite component which accepts the signals coming from the earth station and then shifts that signal to another frequency, once on a new frequency, the

signal is then amplified and re-broadcasted downlink. Satellite frequency bands are grouped as follows [9]:

- a) **C-Band** – downlink at 4GHz and uplink at 6GHz. Fairly tolerant to adverse weather conditions, but shares these ranges with terrestrial systems, which might lead to prolonged discussions for licensing.
- b) **Ku-Band** – downlink at 11GHz and uplink at 14GHz. This band allocation is reserved only for satellites, hence there are no conflicts with terrestrial systems, but can be affected by bad climate conditions.
- c) **Ka-Band** – downlink at 20GHz and uplink at 30GHz. Offers higher bandwidth, and broadband satellites operate in this band. Because of increased bandwidths, they are more suitable for telemedicine, tele-education, tele-surveillance, and networked interactive games than any other satellites operating in other bands.
- d) **L-Band** – operates in 390 to 1,550MHz range and support VSAT networks and mobile communications, including handheld terminals (such as PDAs), vehicular devices and maritime applications. Due to lower frequencies, systems in this band are more tolerant of adverse weather conditions than other systems.

2.2.9.2.2. Applications of Satellites

Several applications of satellites include [9, 16, 17]:

- a) Useful in meteorological and military services
- b) Service remote areas with no terrestrial facilities
- c) Cost-effective provision of point-to-multipoint communications such as International television
- d) Disaster recovery support
- e) Remote monitoring and control
- f) Vehicle tracking and surveillance
- g) Mobile communications
- h) Internet backbones
- i) Used to support telemedicine, tele-education and remote imaging
- j) Location-based applications such as Google Earth.

2.2.9.2.3. Very Small Aperture Terminal (VSAT)

VSAT [9] relies on the use of small terminals, which can be placed outside windows, to provide private network connections to businesses. With VSAT, point-to-point connections are established between two points as a replacement for the expensive leased lines. So, multilocation companies such as banks, universities, and health centers can use the VSAT stations to link their offices in different locations to the central/main office. With VSAT, such companies realize cost savings of up to 50% compared to leased lines. Other applications of VSAT is to provide business video and in disaster recovery measures where land-based

facilities are destroyed. Lately VSAT is used to provide broadband Internet access. Telecom Lesotho (TL) currently offers VSAT services. VSAT systems are easy to deploy, scalable and insensitive to distance-related transmission costs, i.e. transmission cost is the same whether the locations are 150 or 5,000km apart, as long as they are within the footprint of the satellite.

2.2.9.2.4. Advantages and Disadvantages of Satellites

Advantages [9]:

- a) Access to remote areas
- b) Coverage of large geographical areas
- c) Insensitive to topology
- d) Distance-insensitive costs
- e) High bandwidths

Disadvantages [9]:

- a) High initial costs
- b) Propagation delay with GEO systems
- c) Environmental interference problems
- d) Licensing requirements
- e) Regulatory constraints in some regions/countries
- f) Danger posed by space debris, solar flare activity and meteor showers.

2.2.9.3. Cellular Communications

Cellular communications systems rely on a large number of low-power wireless transmitters, called base transceiver stations (BTS) to offer mobile subscribers access to the services (mainly voice, data and currently video). BTSs cover a limited geographical area called a cell, hence the name “cellular communications”. In [19], the cell is defined as: “*the basic geographic service area of a wireless communications system*”. Traditionally, transmitters were high-powered and covered a large geographic area, but with a limited number of subscribers, mainly because the spectrum used for wireless communications is very limited. Due to the increases in the number of users (subscribers) and the need to maximize the use of the spectrum, low-power transceivers were introduced, which allow cells to be sized according to the subscriber density and demand within a particular region. A typical example is the large number of BTSs used in congested urban areas, which cover small areas usually called pico-cells and micro-cells. However in rural and sparsely populated areas, a small number of BTSs, which cover very large areas – called macro-cells, are used. In order to maintain a continuous communication, when mobile users move from cell to cell, their conversations are “*handed-off*” between cells, thus maintaining a seamless service.

2.2.9.3.1. Definition of Terms in Mobile telecommunications

Several terms are regularly used without a clear understanding in telecommunications and this confusion is cleared by defining them as follows [20, 24]:

- a) **Portability** – defines a scenario in which only a terminal is moved and then connected again at another point in the network
- b) **Movability** – when a subscriber moves his personal access, for example, logging onto the data network from different network positions
- c) **Mobility** – complete ambulatory capability in which both the terminal and the subscriber access can be moved with the network automatically keeping track of all movement. Three types of mobility include [20]:
 - Terminal mobility – a characteristic of a wireless network, which allows connections to or from a mobile terminal which is in the service area of the network;
 - Personal/User mobility – a characteristic of a service, which allows a user to make calls from and have incoming calls directed to a user-selected terminal. This user mobility in fixed telecommunications networks is called *universal personal telecommunications* (UPT) [16, 20]. With UPT, services are connected to the subscriber, who has a unique identification (personal number) that is not related to any specific terminal. This

allows a subscriber to use any terminal, identifying himself by the personal number, to make and receive calls and have the cost charged from his personal account [16]; and

- Service mobility – a characteristic of a service to describe the ability of a service to present the same look and feel, i.e. interface, announcements, tones, procedures, to a user irrespective of the terminal and the serving network to which the user is connected.
- d) **Fixed network** – a telecommunications network, using wireline or wireless access, not supporting terminal mobility. Offers limited mobility, whereby a terminal is associated with a particular physical access port of the network
- e) **Wireline network** – a telecommunications network in which users are connected to the network by an electrical or optical link
- f) **Wireless network** – a telecommunications network in which users are connected to the network by means of a radio access method
- g) **Mobile/Cellular network** – a wireless network which supports terminal mobility within a service area and beyond based on the roaming [16, 20] agreements entered into by the local cellular operator
- h) **Wireless local loop (WLL)** – a wireless access network which requires the user to be in the area covered by a specific transceiver.

2.2.9.3.2. Cellular Network Evolution

Cellular communication systems have evolved over the years from the analog first generation (1G), which were deployed in the early 1980's [18, 20]. 1G systems were developed according to standards in individual countries, thus limiting usage within national boundaries, thus completely avoiding economies of scale. In the early 1990's [18, 20], digital second generation (2G) – Global Systems for mobile communications (GSM) were deployed as a replacement of 1G. 2G systems use a two access techniques namely: time division multiple access (TDMA) and code division multiple access (CDMA). TDMA allows up to eight (8) users to share a single 200 KHz radio channel by allocating a time slot to each user – channel is shared based on time. Other 2G systems use code division multiple access (CDMA) – where users on a single channel are distinguished by their unique codes. From 2G the evolution was the introduction of data services in 2.5G, with example technologies such as general packet radio service (GPRS). Due to increased data rates demanded by users, all-IP services in the form of 3G were introduced, with HSDPA as a sample technology, and the future all-IP mobile 4G systems being researched and hoped to provide even more data rates.

2.2.9.3.3. Cellular Communications Network Architecture

Cellular communications architecture for GSM is shown as figure 2.7, which is divided into four distinct stages namely: the user part, BSS, NSS and other networks.

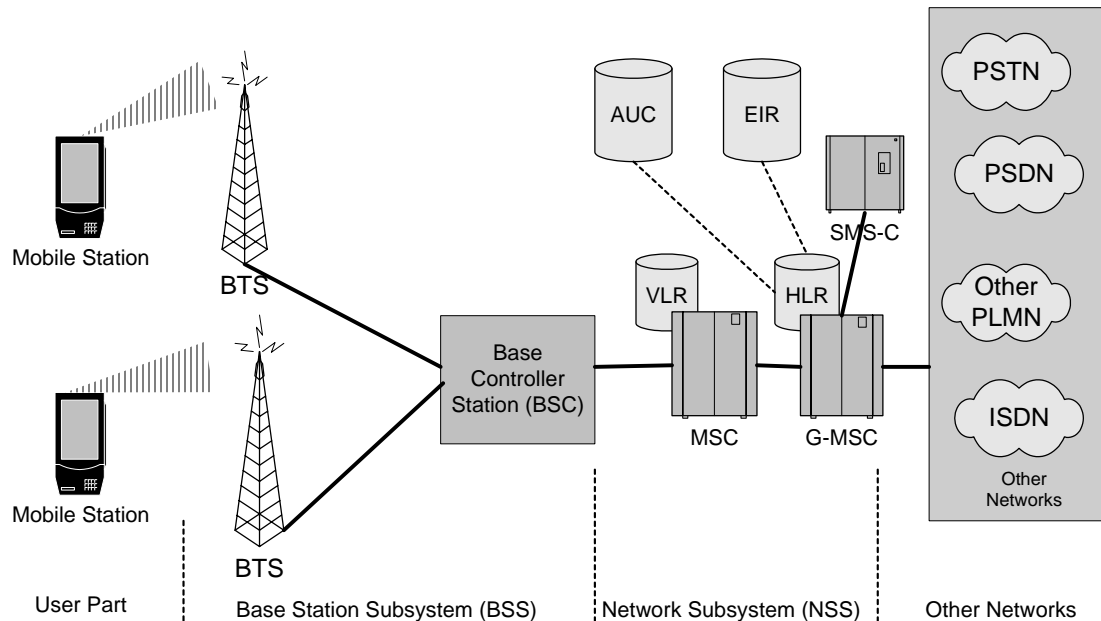


Figure 2.7, adapted from [18, 20, 24], shows a typical GSM cellular architecture

From figure 2.7, mobile stations (MS) consist of the actual physical equipment composed of the radio transceiver, display (LCD), digital signal processor and a smart card, called the subscriber identity module (SIM). SIM provides personal mobility as it can be used with any legal mobile station. Each mobile station uses a separate temporary radio channel to talk to the BTS [18], with each channel using a pair of frequencies one for uplink and one for downlink. To receive satisfactory service, a mobile station has to be in close proximity of the BTS, as radio energy dissipates over distance. Base station subsystem (BSS) [18, 20] provides and manages the radio transmission path between the mobile station and the mobile switching center (MSC). BSS consists of two parts: the base transceiver station (BTS) and the base station controller (BSC). The BTS consists of several radio transceivers (each with their own frequencies), which are the first contact points for the MS wanting to place or receive a call. BSC handles the setup of radio channels, frequency hopping and handovers for a number of BTSs, and also translates the 16kbps voice channels used over the radio link into 64kbps required by the PSTN. The network subsystem (NSS) [18, 20] manages the switching functions of the system and enables the MSCs to communicate with other networks. NSS consists of the actual switches (MSC and gateway MSC) and databases, which are used to store data for user mobility management.

MSCs control calls, tracks billing information, locates cellular subscribers (through interaction with databases), and switches calls to other networks – public switched telephone service (PSTN), public switched data network (PSDN), and other mobile networks – through the gateway MSC (GMSC). Home location register (HLR) [20, 24] – a database located with the GMSC, which contains the permanent customer data on all customers of the mobile network and also contains the identity of the VLR, which knows the whereabouts of each mobile station. Visitor location register (VLR) [20, 24] – a database located with every MSC and has a temporary record of every mobile station logged onto the location area. The short message service center (SMS-C) is used to store text messages used to increase accessibility in a PLMN.

2.2.9.3.4. Mobility management and Authentication in GSM

Switches used in GSM/cellular communications are the same as those used in PSTN, with the exception of the means of user access – provided via BTSs – and the use of special database (HLR and VLR) – for mobility management. To achieve mobility, a large amount of data with logical association – as opposed to physical association in PSTN – is interrogated and used by the mobile network. Before services can be used, the MS has to be first identified; this identification is achieved by a unique number called the international mobile equipment identity (IMEI) [20]. After identifying the MS, it has to be associated with a user profile, which is identified by the international mobile subscriber identity (IMSI) [20], contained in the SIM card. IMSI identifies the customer, a secret key for authentication and other customer information. After the association has been established, the location of the MS in relation to the network's base stations has to be established. Each IMSI has associated with it one or more phone numbers called the mobile subscriber ISDN (MSISDN) number [20]. The IMSI identifies a calling MS to the network and the MSISDN is retrieved and used as a calling line identity (CLI) during call routing. Authentication is provided through two databases namely [20, 24]:

- a) Equipment identity register (EIR) – contains a list of all valid mobile equipments on the network. EIR is used to check that a mobile is not reported as stolen or barred for some other reason;
- b) Authentication center (AUC) – stores security information such as user-specified keys used in authentication of the user. AUC is also used for encryption/decryption.

2.2.10. Hybrid Technologies

Hybrid technologies are made of a mixture of other access media, and the most popular hybrid is formed with fiber and copper called Hybrid fiber copper/coax (HFC). HFC is used mainly in cable television, already discussed under section 2.2.7.2; VDSL and power line communications are other examples discussed next.

2.2.10.1. Very-High-Speed DSL

Information about VDSL has been extracted without alteration from [10], for the purposes of CS5440 course only.

VDSL is a digital data transport technology, which uses a single copper wire pair as a physical medium to provide broadband services, for the last-mile. It is a high data rate next-generation derivative of DSL technologies providing speeds of fiber optic cables and is viewed as bridging the copper telecommunications infrastructure of the past and today with the potentially all-fiber infrastructure of tomorrow. VDSL supports both symmetric and asymmetric data flows, thus providing increased flexibility. Due to its high data rates, VDSL supports several applications including: high-speed Internet access, voice, voice over IP (VOIP), digitally encoded video (requires 6 to 8Mbps), broadcast digital TV, video conferencing, high-definition TV (requires 16Mbps), telemedicine, video games, electronic publishing, etc. Failure of other DSL derivatives to offer video services has led the use of VDSL, which is in direct competition with other broadband access technologies such as PLC, FTTH, etc. VDSL utilizes higher-frequency spectrum (200 KHz to 12MHz) over standard copper above frequencies used for plain old telephone services (POTS) and integrated services digital network (ISDN) services, a technology commonly referred to as data- and video-over voice. VDSL operates over short loops and the extension is provided by installations of fiber optical cables, hence why it's referred to as "hybrid fiber-copper" system. In May 2006 ITU-T approved VDSL2, with improvement in both range and data rates to VDSL at 100Mbps symmetrical and 150Mbps asymmetrical over short copper loops.

Asymmetric VDSL

In asymmetric mode, VDSL offers different data rates for upstream and downstream data flows, with downstream being usually higher. With this mode, VDSL supports asymmetric services, normally called "Class I" services. These services require higher downstream data rates compared to upstream data rates. Downstream involves transmissions from the central office (CO) to the customer premises equipment (CPE); while upstream involves transmissions from CPE to the CO. Asymmetric VDSL data rates are of the order 22-23Mbps downstream and 3-4Mbps upstream for 1km ranges. These rates enable services like [6, 18]: digital TV (DTV), high definition TV (HDTV), multimedia entertainment, super-fast web surfing, file transfers, virtual

office at home, distance learning, telemedicine, video on demand (VOD). For shorter ranges (≤ 300 metres), asymmetric VDSL offers downstream speeds up to 52Mbps, with the possibility of simultaneously delivering multiple DTV or HDTV channels. As an example, HDTV requires 18Mbps for downstream video content, while for upstream it only requires a few kbps for transmission of signaling information (channel changing or program selection).

Symmetric VDSL

In symmetric mode, VDSL offers equal data rates for both upstream and downstream data flows. With this mode, VDSL supports symmetric services, normally called “Class II” services. Traditionally residential customers requested asymmetric services, but are now starting to also demand equal speed for both upstream and downstream. Symmetric VDSL is however viewed as a business services, with 13Mbps speeds in both directions for 1km ranges and up to 26Mbps for shorter ranges (≤ 300 metres). Symmetric services include [6, 18]: video and teleconferencing, online class discussions, and teleconsulting applications.

VDSL Deployment

Typical deployments require VDSL modems at the end of the fiber network and in the customer premises, as shown in figure 2.8.

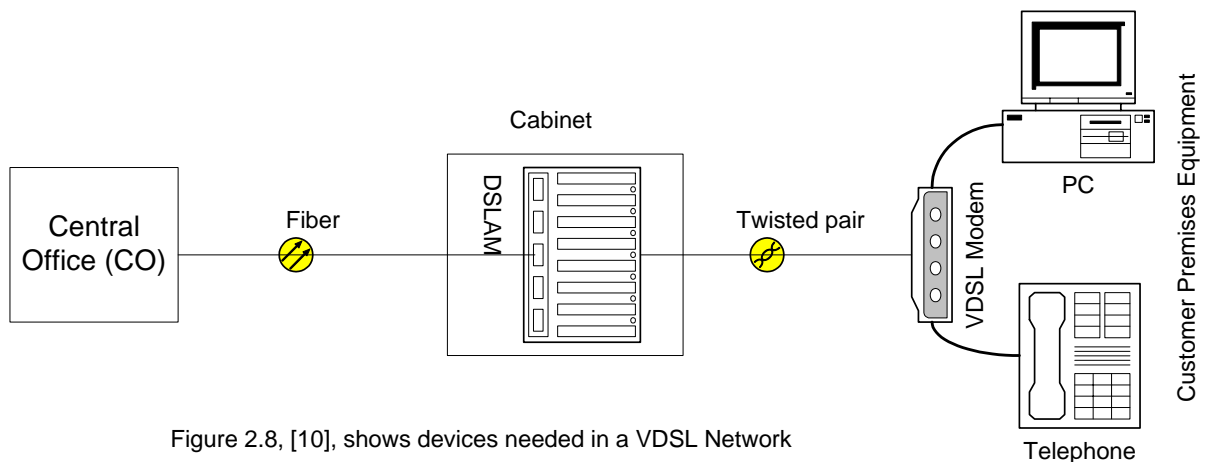


Figure 2.8, [10], shows devices needed in a VDSL Network

The twisted pair wires are connected to the digital subscriber line access multiplexer (DSLAM) within the neighborhood, the DSLAM then links many subscribers through a fiber cable to the central office's main fiber network backhaul. The fiber is terminated by the optical network unit (ONU), which may be independent or integrated within the other end of the DSLAM. The architecture shown in figure 2.1 can be deployed in several ways including the following: Fiber-to-the-neighborhood (FTTN); Fiber-to-the-cabinet (FTTC); Fiber-to-the-exchange (FTTEx); Fiber-to-the-building (FTTB). Any of these deployment strategies can be deployed, and the key to

selecting the most suitable depends on the requirements of the customer and the financial status of that customer.

VDSL Physical Medium

The area from the CO to the CPE is called the subscriber loop and consists of a pair of insulated copper wires of gauges ranging from 0.4 to 0.9mm, with polyethylene as the insulating dielectric. In addition to shielding, copper wires forming a pair are twisted together to reduce interference. These pairs are twisted together in cable binders or bundles each with hundreds or thousands of these twisted pairs. The cable bundles are transmitted together from the CO and are separated at the distribution areas to individual customers either buried or overhead. They are then terminated as wall jacks referred to as “*registered jacks-11*” (RJ-11) inside subscribers’ homes, capable of plugging in a telephone or a modem.

Drivers of VDSL

VDSL supports all broadband applications of today and promises to support the future broadband applications that might be even more bandwidth hungry than the current ones. VDSL is the highest rate DSL derivative and has been forced into action by the following drivers:

- a) Failure of other DSL technologies to offer video services, though “its original function was delivery of video over copper”,
- b) Cost/benefit trade-offs of using FTTC/FTTN as opposed to the full deployment of fiber such as FTTH option, which is very costly,
- c) Re-use of existing copper plant to offer fiber-like speeds, and
- d) Full-service capable broadband residential access networks, with bundled multimedia services.

Benefits of VDSL

VDSL offers fiber like speeds over existing telephone copper cables, with minimal disturbances to the subscribers’ homes and possess several other benefits including:

- a) Provides and supports higher bandwidth services including video, while minimizing expensive plant upgrades,
- b) VDSL architecture has a better security benefit provided by point-to-point links,
- c) “*VDSL provides network access provider additional revenue streams for no additional investment and future proofs the customer*”, and
- d) Bridges the gap between today’s copper world and the anticipated all-fiber network of tomorrow.

Drawbacks of VDSL

Despite the already mentioned benefits, VDSL also has some drawbacks that threaten to undermine its adoption, including the following:

- a) Limited reach, as a trade-off for higher speeds VDSL covers shorter ranges,
- b) The need to extend the loops with fiber cables is costly and involves a lot of labor, time and resources,
- c) Laying of new fiber involves land rights permission and acquisitions, which normally take lengthy discussions with the authorities,
- d) VDSL signals interfere with amateur radio services in close proximity to street cabinets.

2.2.10.2. Power Line Communications (PLC)

Information about PLC has been extracted from [10, 11], without alterations for the purposes of CS5440 only.

PLC identifies all technologies, equipment, applications and services that allow transmission of data over electricity power lines, originally meant for transmission of electricity, and follows the same concept as cable TV and DSL technologies. Electricity uses the frequency of *50Hz* in South Africa and most countries and *60Hz* in United States, leaving the entire spectrum to be exploited. However the frequency used for data must be far away from those used by standard AC so as to avoid interference between the two services. For example band *3 – 148.5 KHz* is for PLC narrowband applications while band *1 – 80MHz* is for PLC broadband applications. Since the power lines were not meant for data, they provide a hostile and dirty last-mile solution, but have the potential for high data rates of *45Mbps* up to theoretically as high as *200Mbps*. With PLC several applications including video on demand, HDTV, entertainment on demand, IP cameras, video conferencing, VOIP, etc are supported. With PLC an IP signal is converted to PLC signal by the head-end device, superimposed over power lines and converted back to IP signals by the modem at the customer premises an vice versa as shown by figure 2.9.

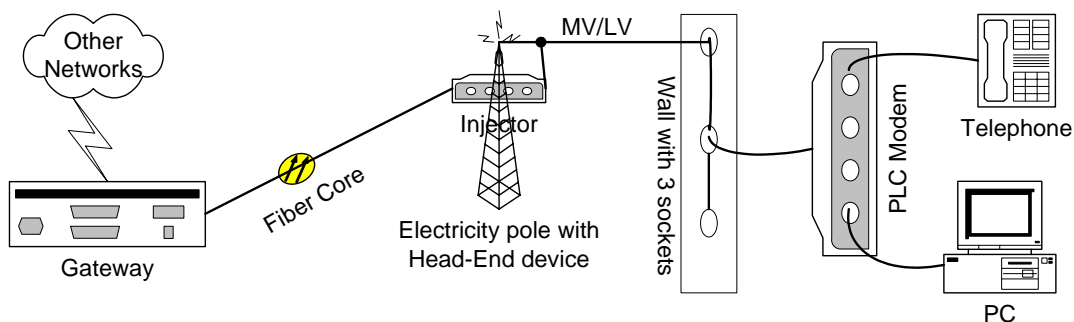


Figure 2.9, [10, 11], shows the devices needed for a PLC network

There are two types of PLC namely: *Access PLC* – all technologies that provide users access to the power lines and broadband services via the use of external electricity grid; and *In-home PLC* – all technologies that provide PLC applications within the home i.e. wall sockets or plugs. With PLC technology, every wall plug socket is turned into a data connection point i.e. electricity and data access coexist on the outlet, thus making in-home networking a real possibility. This design project concentrates on the design of the access PLC network for West Campus Village, and then compares it with VDSL access technology.

PLC Physical Medium

PLC is transmitted over power lines, which extend from the power generation station, all the way to the subscriber's home, as shown by figure 2.10.

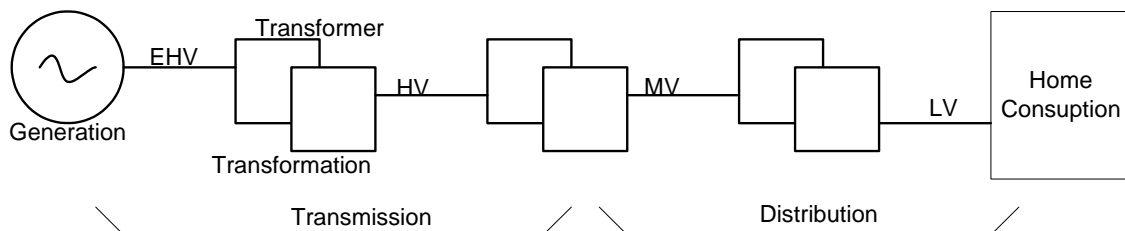


Figure 2.10, [10], shows electricity generation, transmission, distribution and voltage levels from power plant to the consumer

Power generation occurs at power stations, which generate very high voltages (above $36kV$) or even EHV - extremely high voltages (above $300kV$), transmitted to certain areas forming a long-distance nationwide network, mostly overhead, through high voltage (HV) lines. From HV lines, the power is stepped down through the use of transformers located at substations and transmitted over medium voltage (MV) lines to bring electricity into cities, towns and villages. The medium voltage covers ranges from 1 to $36kV$. From MV lines, the power is further stepped down through the use of transformers located close to homes and transmitted over low voltage (LV) lines either overhead or underground into homes for domestic use. The low voltage covers levels below $1kV$. PLC is mainly transmitted over the MV and LV lines, and not over the HV lines. In subscribers' homes, electricity is terminated by sockets located at walls in the buildings, with most buildings having more than one socket in each room as opposed to the telephone jacks that terminate VDSL signals. Power lines conductors are coated by some kind of insulation material to reduce attenuation; this insulation material may be either polyvinyl chloride (PVC) or oil paper insulation, which results in cables suffering lower attenuation than with PVC.

Benefits of PLC

PLC provides high data rates, yet with minimal disruption to the consumer homes during installations and deployment, and has several benefits including:

- a) PLC uses the infrastructure that is already available in design areas thus saving a lot of costs needed if new infrastructure were to be put in place
- b) Very helpful in areas where copper cables have not yet been deployed such as in rural areas,
- c) Allows several services to be offered including: telephony, true video on demand, high-speed Internet, high-resolution picture sharing, video sharing, voice over IP and security cameras.
- d) No need for extra wiring, hence less costs as there is no need for extra points such as for phone and data,
- e) In-house installation is easy and fast, yet PLC offers high bandwidths.

Drawbacks of PLC

Despite the obvious and tempting benefits of PLC technology, there are several drawbacks, which need to be considered before deploying PLC as the broadband access technology of choice in residential and business scenarios. These drawbacks include the following:

- a) The channel is very hostile, unpredictable, accumulates lots of noise, suffers from varying impedance and high levels of attenuation,
- b) There is still a lack of a sound business model for PLC and regulations for PLC,
- c) Some security concerns are raised by the use of PLC, since it uses a shared medium.
- d) PLC signals cannot pass through transformers, hence a need for coupling or bypassing them, which could be costly.
- e) Power lines need repeaters at least every 300 meters and data signals cannot pass through transformers, resulting in a need to bypass them, which could be costly,
- f) PLC has some issues, which are yet to be researched, like the *sky wave* mode of propagation, which has the potential to interfere with radio services at the other parts of the world,
- g) Oppositions from all amateur radio operators worldwide claiming suspected interference with their radio operations.

2.3. Transport Network

2.3.1. Introduction

The transport network in CS5540 is taken as that part of the telecommunications systems, which provides transmission facilities between the provider's network and the access network. The transport network consists of the entire transmission infrastructure and technologies used to carry aggregated or multiplexed traffic. *"The transmission infrastructure allows many communications channels to be carried in multiplexed form between points in the network, usually between switches"* [7]. The transport network is part of the core network, which also contains switching and signaling. The demands on the transport network are immense with the current wave of broadband applications and triple play services. All these demands place greater urgency for the development of technologies capable of multiplexing and effectively carrying the increasing traffic, as evidenced by the following quote. *"The amount of data traffic relative to voice traffic on optical networks and the total traffic volume keeps increasing"* [21]. When designing the transport network, three key requirements have to be considered as follows [16]:'

- a) **Quality** – transferred messages must not be garbled to an extent that is unacceptable to the users, considering what they pay for the service;
- b) **Capacity** – must be dimensioned so that messages will be transferred without being delayed or blocked in the network. In addition, this capacity must be scalable enough to accommodate the daily increases in subscribers and traffic carried; and
- c) **Diversity** – if a fault occurs in the path normally used for the transfer of a message, an alternative path should be immediately made available. This is the robustness or reliability of the network to bypass failures.

The transport network serves as the carrier of the bearer networks, such as PSTN, PLMN, ISDN, PSDN, etc. The network operator has to meet the basic requirements of the services offered while keeping the cost at an acceptable level, i.e. solutions must be economically viable [16].

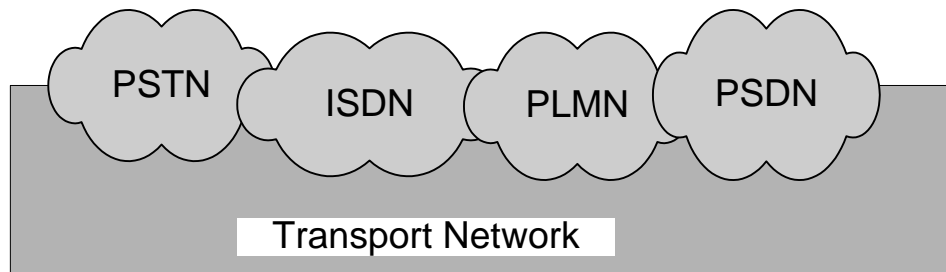


Figure 2.11, adapted from [16], shows transport network as the carrier of bearer networks

Transmission infrastructure and technology are changing rapidly and key trends are [16]:

- a) **Digitization** – from analog techniques to digital ones;
- b) **Fiber** – introduction of fiber optical transmission as the main stream.

These two trends have resulted in reduced transmission costs per channel per kilometer and improved transmission quality. The reduction of transmission costs has led to the declining of costs for long-distance services, hence low tariffs for long distance calls in relation to local calls. This concept is called “*death of distance*” [25]. Several media are used to offer the transport services, such as satellite technology, microwave (terrestrial systems) and fiber optical cables. Only technologies based on fiber will be discussed next, as fiber is the envisaged medium for all future telecommunications.

2.3.2. Fiber-based transmission techniques

Fiber is the medium of the future and the only future proof, truly broadband capable transport medium, expected to carry aggregated future unknown applications. Despite this, other techniques relying on microwave and satellite still have to be considered as alternative during disasters, such as Tsunamis and terrorism acts, when fiber cables are cut or unoperational. The current wave of is seen in several projects aiming to build a fiber optic sea cable from South Africa to Europe, India and the Middle East. Four projects are:

- a) The East African Marine Cable System (TEAMS) Project;
- b) East African Submarine Cable System (ESSAY);
- c) Seacom; and
- d) Reliance.

2.3.2.1. Plesiochronous Digital Hierarchy

As has already been discussed, traffic in the transport network is aggregated or multiplexed from several individual channels (usually 64kbps). This aggregation is done in terms of hierarchies. The term “hierarchy” is used in its usual meaning of a ranking of entities in grades, orders or classes, with one above the other and where each level deals only with the levels immediately above and below it. In the early 1950’s [19], a single copper pair was used to carry only one conversation, with one pair for each direction. As traffic increased due to more subscribers joining the network, this method became very unscalable. To improve upon this, ITU-T recommendations standardized two (2) systems for the multiplexing of several signals on a single physical path. These systems are:

- a) E1 – 30 channels were multiplexed and is mostly used in Europe and the rest of the world, at the maximum rate of 2.048Mbps; and

- b) T1 – 24 channels were multiplexed and is mostly used in North America and Japan, at the maximum rate of 1.544Mbps.

The multiplexing of signals over a single channel is made possible by the digitization of analogue signals and the use of time-division multiplexing [19]. Multiplexing is used to reduce costs, with several channels along the same route sharing the same transmission medium – such as fiber. With further increases in the traffic, there was a need for more channels than the 24 or 30, to be multiplexed. The response was the introduction of “plesiochronous digital hierarchy” (PDH) [19, 25]. The term “*plesiochronous*”, simply means that the signals to be multiplexed run at almost the same speed [25], i.e. their bit rates must be within a specified range of the nominal value [19]. With PDH [19, 25], multiplexing first takes place at the 24-channel (North America and Japan) or 30-channel (Europe and rest of the world) level. Then several of these multiplexed channels are combined to form higher order multiplexes, in the hierarchical form. This process is repeated until higher order multiplexes are achieved, with ITU-T recommendations specifying four (4) levels of multiplex. Table 2.1 shows the different multiplex levels as used in Europe.

Standard or Country	Multiplex level	BIT Rate	No. of 64kbps channels
Europe	E0	64kbps	1
	E1	2.048Mbps	30
	E2	8.448Mbps	120
	E3	34.368Mbps	480
	E4	139.264Mbps	1920
Table 2.1, adapted from [16, 19, 25], shows European PDH hierarchy			

2.3.2.1.1. Limitations of Plesiochronous Digital Hierarchy

PDH suffers from several weaknesses as follows [16, 19, 25]:

- Different hierarchies are used in various parts of the world, thus leading to problems of international interworking, because of this existence of regional standards
- Insufficient capacity for network management
- Most PDH network management is proprietary and very limited
- There is no standardized definition of PDH bit rates greater than 140Mbps
- With increasing complexity of systems and widespread digital switching, the lack of synchronization between the clocks has become a major consideration in system design
- PDH was oriented towards point-to-point links, thus the need for cross-connects, which are not part of PDH standards, to accommodate add/drop of channels along the way
- With PDH, an individual 64kbps channel is used as the point of access, and thus, it's necessary to demultiplex down through all levels of the

hierarchy in order to extract a 64kbps channel. With this inflexibility, PDH is suited for point-to-point links and cannot add and drop channels without demultiplexing to the lowest level. This property of PDH multiplexing is called the “*multiplexer mountain problem*” [19]. This inability to add and drop channels without demultiplexing to the required level proved to be a serious obstacle as more flexible management of the network involving reallocation of multiplexed links to adapt to changes in traffic and fault conditions became essential.

2.3.2.2. Synchronous Digital Hierarchy

PDH was developed as a response to 1960's [16, 19] transmission needs, and was developed within the constraints of available electronic technology. The key aim was for point-to-point transmission links, but with increases in traffic over the 1980's, there was a need for a different technology to replace PDH. High traffic demands meant that links able to support more than terabytes of data were needed. Those high data rates would be made possible by fiber optic cables, which have a potentially unlimited bandwidth. This new technology standard is “synchronous digital hierarchy” (SDH) [19, 25], formulated by ITU-T as a standard for telecommunications transport. SDH was first introduced into telecommunications networks in 1992 [25], and has been deployed at rapid rates since then. SDH is based on overlaying a synchronous multiplexed signal onto a light stream transmitted over fiber-optic cable. The most important feature of SDH is the introduction of manageability, which facilitates [16]:\

- a) Centralized remote control of network elements;
- b) Increased use of the physical network; and
- c) Shorter delivery time for leased lines.

With this introduction of SDH, the transmission network was then termed the transport network, to denote the added manageability. SDH defines synchronous transport modules (STM) for the fiber-optic based transmission hierarchy as shown in table 2.2. STMs range from zero (0) to N – as in STM-0 to STM-N. STM [16, 19, 25] is an information structure consisting of an information payload, together with section overhead information that supports connections at the section layer.

Multiplex Level	Bit Rates in Mbps	SDH Capacity
STM-0	51.84	21 E1
STM-1	155.52	63 E1 or 1 E4
STM -4	622.08	252 E1 or 4 E4
STM-16	2.4 Gbps	1008 E1 or 16 E4
STM-64	10Gbps	4032 E1 or 64 E4
STM-256	40Gbps	16128 E1 or 256 E4
Table 2.2, adapted from [16, 19, 25], shows SDH hierarchy		

2.3.2.2.1. Terminology used in SDH

Three key terms used in the transport network are defined as follows [16, 19, 25]:

- Synchronous – digital transitions in the signals occur at exactly the same rate;
- Plesiochronous – two signals are said to be plesiochronous, if their transitions occur at almost the same rate, with any variations being constrained within tight limits; and
- Asynchronous – transitions of the signals do not necessarily occur at the same nominal rate, i.e. the difference between two clocks is much greater than a plesiochronous difference.

2.3.2.2.2. Requirements on SDH

As a replacement of PDH, SDH has several requirements to satisfy including [19]:

- Be synchronous
- Be service-oriented as opposed to point-to-point orientation of PDH
- Allow multiplexes of any level from 24 and 30 channels upwards to be inserted or extracted and mixed in any number of ways thereby providing a switching function at the multiplex level
- Be highly manageable and provide for transmission of significant management data across its links
- Provide high data rates, selected initially as 155Mbps and 622Mbps, with the intention to move upward in four-fold steps, limited only by availability of technology
- Give reduced bit error ratios – reduced from the level of 1 error in a million bits in PDH to 1 in 100 million – a performance made possible by optic fiber.

2.3.2.2.3. Advantages of SDH

Benefits of SDH include [25]:

- a) A reduction in the amount of equipment and an increase in network reliability
- b) The provision of overhead bytes (used for management of payload bytes on an individual basis) and payload bytes
- c) The synchronous multiplexing format simplifies interface to digital switches, digital cross-connects and add-drop multiplexers
- d) Availability of a set of generic standards, which enable multi-vendor interoperability
- e) The definition of a flexible architecture capable of accommodating future applications with a variety of transmission rates
- f) Multipoint configurations are supported.

2.3.2.2.4. Network elements in SDH and its Architecture

Network elements in SDH are divided into [16, 19, 25]:

- a) Digital Cross-Connects (DXC) – terminates a number of SDH links and rearranges the connections to meet variations in the needed capacity in normal operation and to perform “protection switching” in the case of a cable break;
- b) Multiplexers – there are two types namely:
 - i. Add/Drop (ADM) – can insert (add) or remove (drop) and part of the payload to or from an STM-N
 - ii. Terminal (TM) – multiplexes and demultiplexes SDH payload from a common STM-N.
- c) Regenerators – called the intermediate repeaters (IR), are needed to regenerate (recover timing) and restore the signal level, when distances between multiplexers is significant.

To facilitate better management, SDH networks are grouped based on the geographical setting from international, national up to metro networks as shown in figure 2.12.

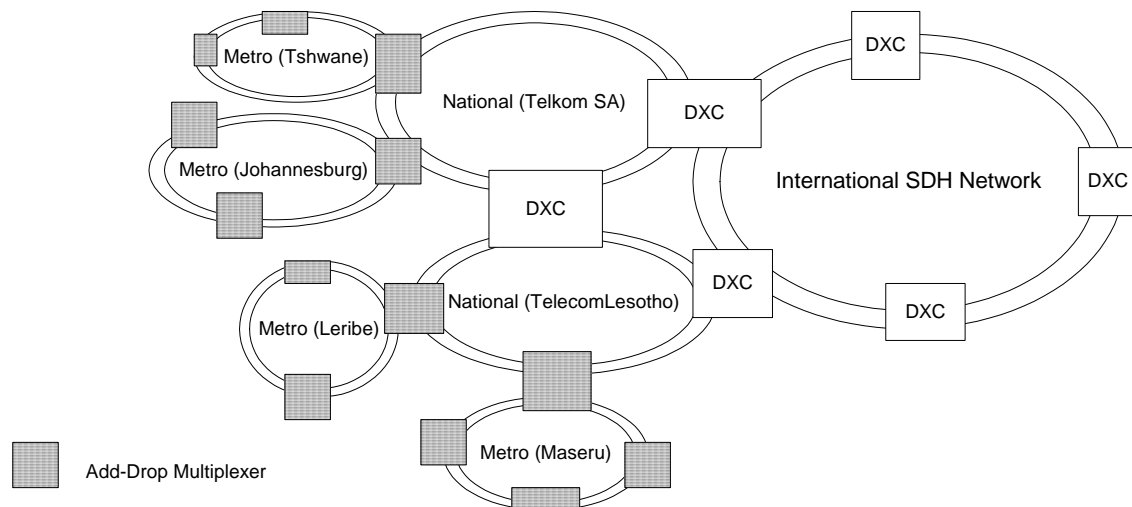


Figure 2.12, adapted from [16, 19], shows physical SDH network architecture from International to national to metro areas

2.3.2.3. Synchronous Optical Network

Synchronous Optical Network (SONET) [16, 19], is the same standard as SDH, except for limited features, used in North America. SONET is a fiber-optic transmission protocol that provides synchronous transport by means of a very clear and simple scheme, and is used primarily for network transport of broadband communications (51.84Mbps) [19] or greater between and among carrier switching nodes. SONET, just like SDH was developed for the following purposes [24]:

- Synchronous fiber-optic networking
- Efficient add/drop multiplexing
- Compatibility of equipment among manufactures of equipment
- Robust fiber rings - survivable fiber-optic rings, which can tolerate a single cut in the fiber path
- Support of new services such as ATM
- Enhanced network management capabilities - called OAM&P for operations, administration, maintenance & provisioning.

SONET uses synchronous transport signal (STS) [16] as an equivalent of SDH's STM, as shows in table 2.3. To recognize the equivalence of SONET and SDH, most references refer to them as one technology, using "SONET/SDH" [21] notation.

SONET	Bandwidths (Mbps)	SDH Equivalence
STS-1	51.84	STM-0
STS-3	155.52	STM-1
STS-9	466.56	
STS-12	622.08	STM-4
STS-18	933.12	
STS-24	1244.16	
STS-36	1866.24	
STS-48	2488.32	STM-16
Table 2.3, adapted from [16, 24], shows SONET hierarchy compared to SDH		

2.4. Switching

2.4.1. Introduction

Switching and transport network together for what is called a “core network”. **Section 2.3** briefly discusses the transport network, which is composed of all the transmission infrastructure and technologies. This section briefly discusses switching. Switching can be defined as the act of setting up connections between two subscribers, so as to enable communications (sharing and transfer of messages) to place, as shown in figure 2.13.

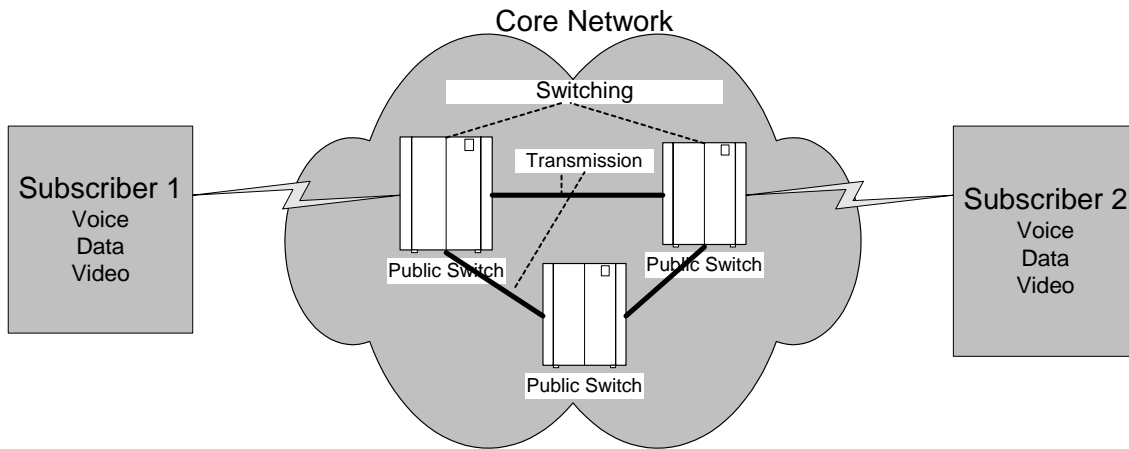


Figure 2.13 [16], shows switching in the telecommunications network

ITU-T has defined switching as follows [16]: “the establishing, on demand, of an individual connection from a desired inlet to a desired outlet within a set of inlets and outlets for as long as is required for the transfer of information”. Information in this case includes, voice video and data (triple play). In addition to the user information, there is also information needed by the network – known as signaling information – which has to be switched as well. Switching improvements/developments are driven by three (3) key factors as follows [16]:

- a) Accessibility or ability to establish desired connections – switching technology must enable the operator to be able to establish a successful and chargeable connection from the sender to the receiver through the network. Dimensioning of switching capacity is one of the important design factors, so as to avoid congestion. Reliability of the equipment must also be considered.
- b) Transparency – ensures that delays through the switching equipment are minimized and that flow of information is not distorted in any way and that the switched bandwidth can match service requirements.

- c) Network economy – switching equipment must be capable of handling multiplexed traffic that consists of packets. Packet-mode switching techniques were introduced in a bid to use lines more efficiently.

2.4.2. Switching Techniques

Traditionally only circuit switching was used, which is very suitable for isochronous services, such as voice telephony. Circuit switching [22], defines a technique whereby a fixed circuit is setup end-to-end before communications can occur, and is maintained during the period of use, and then finally torn down (released) once conversation is over between the communicating parties. Communication via circuit switching involves three (3) stages as follows:

- a) Circuit establishment;
- b) Information transfer; and
- c) Circuit termination.

Circuit switching provides a sort of a dedicated connection, once setup has occurred and shares the benefits of a dedicated connection as follows:

- a) 100% bandwidth is available to end-users for the duration of the call – dedicated connection;
- b) Guaranteed data rates;
- c) Delay is limited to propagation time;
- d) No need for data reassembly;

Despite the benefits, there are however several downfalls of circuit switching including:

- a) Delays prior to call establishment
- b) Very expensive, especially when used moderately per unit time
- c) Very inefficient as the channel capacity is dedicated for the duration of the call even if no data is transmitted, e.g. in speech where several pauses are the norm
- d) Not suitable for the transfer of data packets, hence in the converged world of today, circuit switching has no future.

In recognition of these drawbacks, subscribers demands for better utilization of transmission capacity, demand for larger bandwidth and the emergence of data communications, packet switching was introduced, in the 1970's [16] as an alternate switching technique. Packet switching has the following characteristics:

- a) Data is first small separate blocks called packets
- b) Each packet must have a destination address
- c) Packet delivery is random and reassembly at the receiver's side is needed

- d) Designed for asynchronous communication, with each packet being individually addresses and routed
- e) Bandwidth is shared by all users connected to a network resulting into more users, but less bandwidth per user (*advantage as well as a drawback*)
- f) Each time a packet passes through a packet switching node, it is subjected to delay not present in circuit switching
- g) When traffic becomes heavy on a circuit switched network, some calls are blocked, but with packet switching, packets are still accepted but delivery delay increases.
- h) Communication capacity for carrying user traffic is reduced due to overhead information including destination address and sequencing information that must be added to each packet

Delay in packet switching is too much, because at each node on the route, the packet is received, stored briefly and then passed on to the next node. In addition, delay due to processing and queuing in the switching nodes. Minimum delay is given by equation 2.1 as follows:

$$MinimumDelay = \frac{L}{R}$$

Where: L – length of packet in bits

R – Incoming channel rate in bps

Figure 2.14 shows different switching techniques, including frame-relay and cell switching.

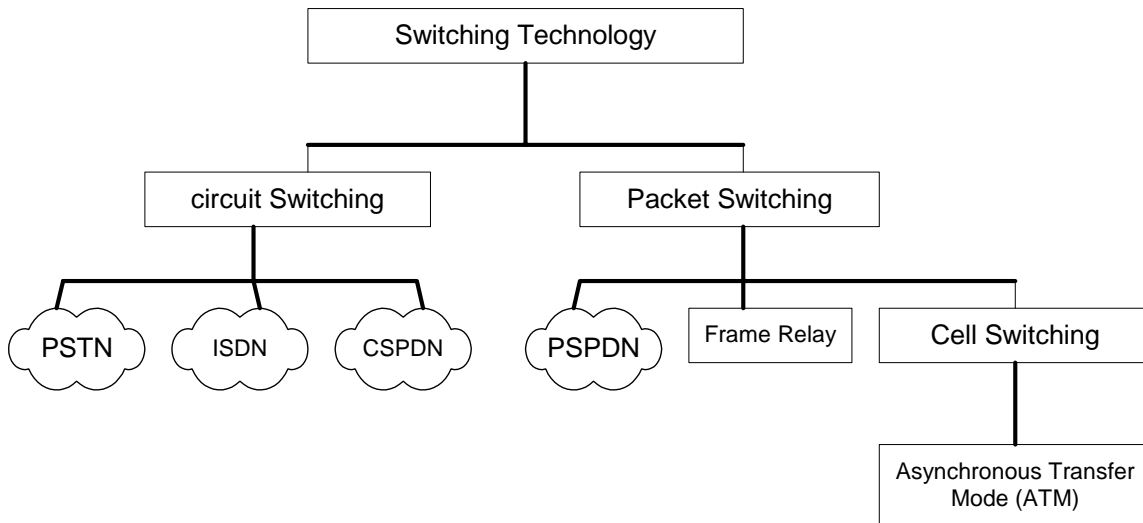


Figure 2.14 [16], shows different switching techniques in public networks

2.4.3. History of Switching

In 1878 [16], manual exchanges [16, 19] were implemented in the UAS. With this manual exchanges, the human operator received calls and switched them manually to the called subscriber – the operator sets up a circuit between two subscribers – hence the expression “circuit switching”. When the conversation was complete, the human operator released the connection.

From manual exchanges, the advancement was the introduction of electromechanical [16, 19] systems in the early 1900’s [16]. With these systems, the number of human operators was significantly reduced – thus cutting labor costs – as they could select switching paths by analyzing the information provided during dialing; though they required more maintenance.

From 1960’s [16] digital, computer-controlled systems were introduced. These systems included telephone exchanges, which also had analogue-to-digital (A/D) converters and were fully computerized. Due to the increased use of data communications and increased data traffic, special nodes for data communications were also introduced. The need for integrated traffic (voice, video and data) in the form of ISDN, also led to the introduction of switching nodes for N-ISDN. ISDN nodes are a combination of a telephone exchange, packet data switch and an important sorting function for subscriber traffic. B-ISDN nodes were made available with the introduction of ATM cell switching and there are now optical switches, which match the speeds offered by fiber optical cables, so as not to limit the bandwidth of a connection.

2.4.4. Nodes used in switching

As has been discussed, switching applies to both voice (PSTN) and data (PSDN) networks, but uses different techniques. The concept of switching whether used in PSTN or PSDN simply involves the means of routing individual channels [19]. A channel is defined in [19] as: *“the path through which the speech signals are transmitted from sender to receiver”*.

2.4.4.1. Nodes used in circuit switching

Circuit switching uses telephone exchanges (PSTN) exchanges, shown as figure 2.15.

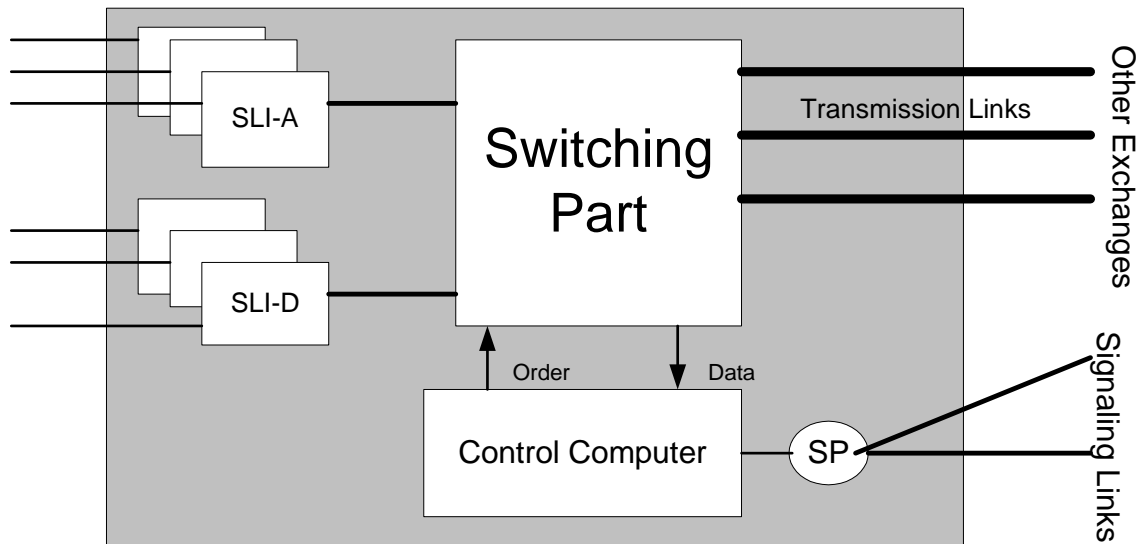


Figure 2.15, adapted from [16, 19], shows the design of a telephone (PSTN) exchange

From figure 2.15, the following distinct parts are identified [16, 19]:

- Access part – provided by the use of printed circuit cards called subscriber line interfaces (SLI), for both analogue (A) and digital (D).
- Switching part – based on time or space switching, its main task is to interconnect an incoming time slot and an outgoing time slot.
- Control part – a computer housing the switching logic and other programs.
- Signaling point – provides the exchange's control computer with access to SS7 network.
- Transmission links – used for the actual transmission of traffic from exchange to exchange, and have to conform to a PDH or SDH standard.

2.4.4.2. Nodes used in packet switching

Nodes used in packet switching can either be bridges, switches or routers, and the generalized architecture of a packet switch is shown as figure 2.16.

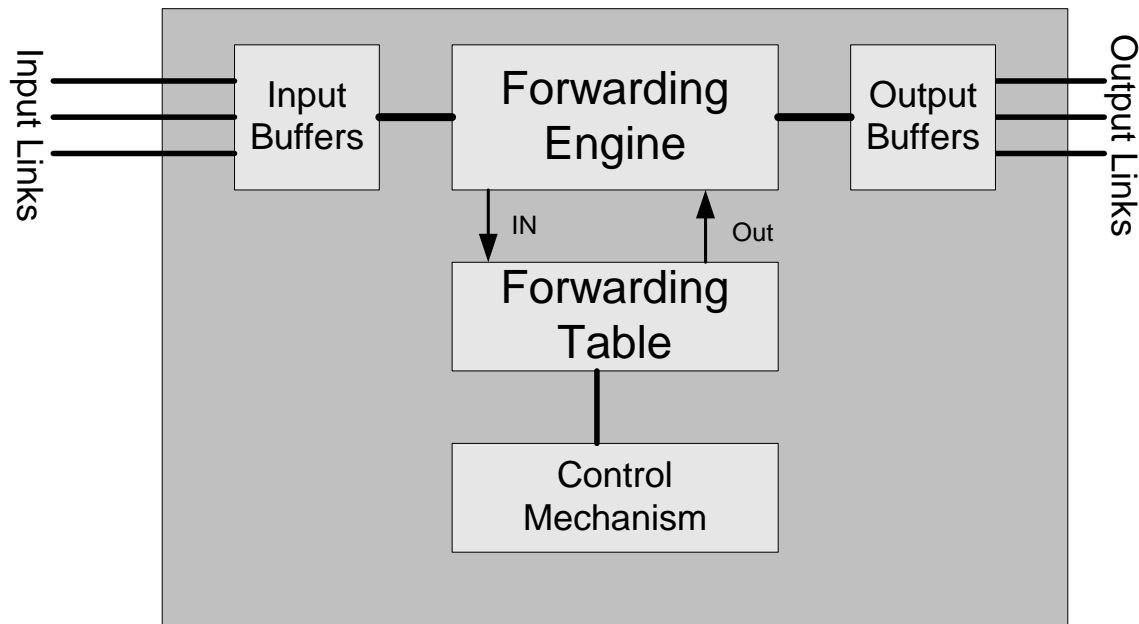


Figure 2.16, adapted from [16, 19], shows a generalized architecture of a packet switch

From figure 2.16, the following parts are identified [16, 19]:

- a) Forwarding Engine (Switching part) – used to perform a number of data link layer processes as follows [19]:
 - i. Input frame processing – detects frame boundaries and checks frame integrity;
 - ii. Address Extraction – extracts the destination address and pass it to the forwarding table;
 - iii. Packet classification – implements different classes of service and shows how to handle packets from each class during congestion;
 - iv. Switching – the actual direction of packets arriving on an incoming link to the outgoing link that leads to the next intermediate system towards the packet destination; and
 - v. Output frame processing – insertion of checksums – if the frame content has changed – and reinserting the address field, when required.
- b) Forwarding table – contains information used in the particular protocol to switch frames. Address information in the arriving packet is used as the index into the table. The table content contains at least the identity of the output link to be used. The layer at which forwarding takes place is determined as follows [19]:
 - i. Layer 3 forwarding – usually termed “routing”, uses the destination address information contained in layer 3 protocol data unit (PU)

- ii. Layer 2 forwarding – usually termed “switching”, uses destination address or virtual circuit information contained in the layer 2 frame.
- c) Control mechanism – routing protocol used or any other process to determine the content of the forwarding table.
- d) Input buffers – store arriving packets in order to give the processors enough time to process the packets – read addresses, etc – when bursts of data arrive at the node.
- e) Output buffers – needed to store a copy of each packet in case of a request for retransmission.

Based on the forwarding modes, two key intermediate networking devices are defined as follows [19]:

- a) Router – a node for directing packets to the next node toward their destinations using network-wide addresses contained in layer 3 PDUs, using information on available routes, their states and costs that is built up by router-to-router information exchange. Hence why it’s used for best path finding through the network. A router is found at the point of ingress to a network and at the gateways between networks.
- b) Switch – a node for directing packets along a pre-determined path toward their final destinations using addresses peculiar to source-destination pair normally contained in the layer 2 frame header and routing information related to end-to-end route. A switch is a much simpler and faster device than a router, and is found in local area networks (where they map IP addresses to MAC addresses), wide area network or in core networks.

2.4.5. Addressing in Telecommunications

Addressing is very useful in communications, because without addresses (identifiers), there would be no communications. The same phenomenon used to give people names, applies to telecommunications as networking devices also need to be uniquely identified, so as to enable unambiguous communications. Addressing uses different formats and standards in both data and voice networks, but the convergence of technologies favors the adoption of addressing as used in data networks. In terms of switching/routing, none can take place without the identification of the devices. So for switching/routing to take place, all devices (host and intermediate) in the network must have unique identifiers.

2.4.5.1. Addressing in Voice networks

Telephone sets in the world are all identified by unique numbers or addresses. The uniqueness is ensured by adopting a standard method of allocating numbers based on conventions for numbering to identify the country in which the address resides and within the country. A convention for allocating the different parts of telephone numbers is called a “numbering plan” [16, 19]. The main purpose of

the numbering plan is to give each subscriber and each service a unique and simple code which makes automatic call set-up possible, with different bearer networks having different numbering plans. The telephone numbering plan is defined as “ITU-T Recommendation E.164 numbering plan [16, 19]. The structure of E.164 telephone number is shown as figure 2.17.

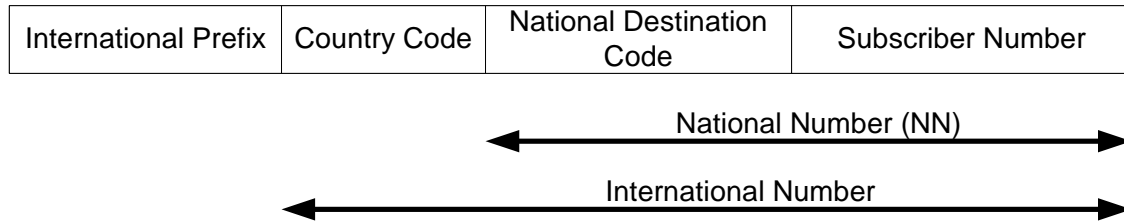


Figure 2.17, adapted from [16, 19], shows E.164 telephone number structure

From figure 2.17, the following are identified [16, 19]:

- a) International Prefix – precedes the international number and simply allows one to call international networks and is normally a single digit, mostly a zero (0). The use of “+” sign simply indicates “*insert your network’s international access code*”. This is not part of the telephone number.
- b) Country code (CC) – a one to three digit number specifying the country or group of countries as follows:
 1. Lesotho – 266
 2. South Africa – 27
 3. United Kingdom – 44
 4. USA, Canada and Caribbean States – 1
 5. Zambia – 260
- c) National Destination Code (NDC) – may take on different forms as follows:
 - Trunk Code (TC) or Area Code (AC) – shows a specific location based on geographic demarcations, examples:
 - i. 34 – Roma
 - ii. 31 – Maseru
 - iii. 11 – Johannesburg
 - iv. 51 – Bloemfontein
 - Destination network (DN) code – does not rely on geographic demarcations and is used in mobile networks as follows:
 - i. 58 – Vodacom Lesotho

- ii. 62/63 – Econet
- iii. 72 – Vodacom South Africa

d) Subscriber number (SN) – used to distinguish subscribers within a certain area or within a mobile operator’s network; example “314 4824”.

The combination of NDC + SN makes up the national number (NN), which is typically 9 to 11 digits. The total length of an international number (CC+NN) is 12 digits but could expand to 15 digits. Fixed network numbers are based on the geographic locations, while mobile numbers are not based on locations. Example telephone numbers:

- i. 266 2234 0601 – Roma
- ii. 27 11 716 5470 – SA

2.4.5.2. Addressing in Data networks

As opposed to E.164 numbering plan used by voice networks, data networks used internet protocol (IP) addresses for host and intermediate device identification. IP is an internetwork protocol; meaning it provides a communication system that works across linked (interconnected) networks. In an internetwork, the individual networks that are joined are called subnetworks or subnets [17]. The interconnection device is usually a router, but bridges can also be used. LANs use different technologies – Ethernet, token ring FDDI, etc – which means that their medium access control (MAC) methods are different, so for communications to occur between such networks, IP is used as a universal way of packaging information for delivery across network boundaries. IP is a connection-less, best effort delivery protocol, which means that packets can and do get lost along the way towards their destinations, but it’s up to the destination end-system to notice and recover lost packets.

Hosts are identified in four ways in the TCP/IP network, as follows [17, 19]

- a) Physical Address – is the MAC address that is hardwired into network interface cards. It is used for LAN addressing, not internetwork addressing;
- b) Network Address - identifies a specific host on an IP internetwork at layer 3;
- c) Domain name - provides an easily recognized name for a host on an IP internetwork. While humans use domain names, they are resolved into IP addresses by DNS (Domain Name System) for general addressing on IP internetworks; and
- d) Socket – an identifier for a particular application, which uses a host computer’s network address and a port number.

IP addresses are numerical addresses that are grouped as four (4) bytes or octets, in the dot notation and to a total of 32 bits. These are technically called IPv4

addresses. Each IP address has a network portion (identifying a network or group of computers) and a host portion (identifying a specific computer on the network). IPv4 addresses (or address space) are grouped into three (3) useable classes namely:

- a) Class A – the first octet ranging from 1 – 126.
- b) Class B – the first octet ranges from 128 – 191
- c) Class C – the first octet ranges from 192 – 223.

IPv4 has been a great servant of the Internet community so far, but the exponential increases of end systems (Hosts) on the Internet, has seen a key limitation of IPv4, which has a very limited address space. Several approaches to alleviate this shortage include:

- a) A solution was developed with the creation of CIDR (Classless Interdomain Routing), which allocated class C addresses as variable-size blocks. A block is a range of addresses (without excess addresses) appropriate for an organization's needs. This leaves addresses free for other users. Still, CIDR only buys time. [17]
- b) Use of private network address, whereby a private address is used within the private network and is then translated into a public address whenever the user needs to communicate with the outside world. This is achieved through the use of network address translations (NAT).
- c) Adoption of IPv6, which is the “next generation” suite of protocols that is intended by IETF to replace the IPv4 protocol suite that currently governs the form of addresses used to identify hosts on the Internet worldwide [17, 28].

The most important feature of IPv6 is its longer address space. It is 16 bytes (128 bits) long [17, 28], compared to 4 bytes (32 bits) for IPv4! That will provide enough addresses to assign an IP address to every person and every conceivable device on the planet. Imagine your home entertainment system has an IP address. From your office, you could send a command to have it record a television show. IPv6 addresses are usually written in hexadecimal notation as a sequence of eight (8), 16-bit blocks separated by colons [28].

Example: 2001:0548:F9A6:0000:00E5:AF02:8C65:2D2F

Several drivers of IPv6 include [17, 28]:

- a) Depletion of the IPv4 address space
- b) Re-invention of secure, end-to-end communications
- c) Auto-configuration
- d) Improved addressing
- e) Improved header structure
- f) Support for Mobile IP

2.4.6. Asynchronous Transfer Mode

Asynchronous transfer mode (ATM) [16, 17, 19], is a form of packet-switching technology, as shown in figure 2.14, that transmits fixed-length units called cells at very high speeds. ATM is not just a switching technology; it is also a transmission technology, which is connection-oriented, thus combining the reliability of circuit switching with the efficiency of packet switching, giving the user the best way to deliver all types of data. It acts as a dedicated, full-duplex switched technology that adjusts bandwidth to whatever users' applications needs are. ATM offers more than just a transmission medium, it also:

- a) supports both private and public networks
- b) uses the same technology for LANs and WANs
- c) transports voice, video and data traffic on a common circuit
- d) delivers bandwidth on demand
- e) offers simplified networking structure

ATM can provide residential subscribers with services like video-on-demand (VOD), telephony and broadband Internet access. ATM can also be used for LAN interconnect, video conferencing, private branch exchange (PBX) interconnect and broadband Internet access.

2.4.6.1. ATM Cells

ATM uses fixed size blocks of data called "cells"; which are 53 bytes long and divided into two as follows [16, 17, 19]:

- a) Header – 5 bytes long and used as an addressing mechanism, carrying routing information based on virtual path or virtual circuit identifiers (VPI or VCI), which define how the cell is to be switched.
- b) Payload – 48 bytes long, is the portion that carries the actual information (voice, video and data).

By using a payload of 48 bytes for data, ATM offers a compromise between a large cell size (64 bytes) for data and a small cell size (32 bytes) for voice. This fixed cell size reduces the queuing delay for high-priority cells during congestion. Another benefit is that small cells meet the low delay requirements necessary for voice or telephony, as well as data services with an acceptable overhead at the same time. Also the small, constant cell size allows ATM equipment to transmit video, audio, and computer data over the same network.

2.4.6.2. ATM Virtual Channels and Virtual paths

ATM is a packet switched technology, which is however connection-oriented, hence before any data can be transferred, logical channels have to be setup between pairs of end-stations. These logical channels or connections between end stations are called virtual channels (VC) [16, 17, 19], and are identified by virtual

channel identifiers (VCI). A new VCI is assigned whenever a virtual link is switched. A VC is setup between two end users through the network and a variable-rate, full-duplex flow of fixed-size cells is exchanged over the connection. VCs are also used for user-network exchange (control signaling) and network-network exchange (network management and routing). A virtual path (VP) [16, 17, 19] is a bundle of virtual channels that have the same end-points. These VCs are thus switched together and are identified by the same VPI. The VPs are then bundled and transmitted through the provided transmission path, as shown in *figure 2.18*. The use of VCs and VPs simplifies the network architecture, increases network performance and reliability, and reduces processing and connection setup time. Once VCs are established, routing tables are created that route the cell to the appropriate output link. Data is then broken into fixed-sized cells at one point, transmitted to the destination as quickly as possible over the VCs and then reassembled in the original order. ATM creates a fixed channel or route between two points whenever data transfer begins. This differs from TCP/IP in which messages are divided into packets and each packet can take a different route from source to destination. This difference makes it easier to track and bill data usage across an ATM network, but makes it less adaptable to sudden increases in network traffic.

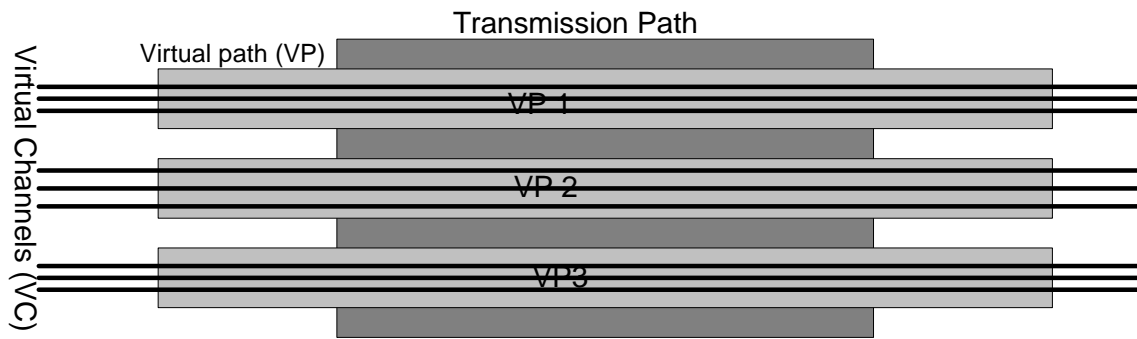


figure 2.18, [16, 17, 19], shows ATM's VC, VP and the transmission path

2.4.6.3. ATM Class of Service

ATM has four classes of service as follows [16, 17, 19]:

- a) Constant Bit Rate (CBR) – requires that a fixed data rate be made available by the ATM provider. The network must ensure that this capacity is available and also polices the incoming traffic on a CBR connection to ensure that the subscriber does not exceed its allocation. CBR provides a similar service to a leased line. This is referred to as “Class A” quality of service and is suitable for the transmission of fixed-rate uncompressed audio.
- b) Variable Bit Rate (VBR) – offers flexibility rather than using a single rate. VBR is defined in terms of a sustained rate for normal use and a faster

- burst rate for occasional use at peak periods. This is referred to as “Class B” quality of service and is mainly suitable for video conferencing.
- c) Available Bit Rate (ABR) – provides a user with a guaranteed minimum capacity. When additional capacity is available, the user may burst above the minimum rate without the risk of cell loss, as bandwidth is adjusted based on the amount of traffic in the network. This is referred to as “Class C” quality of service and is intended for applications in which delay is a concern, such as online sessions between a client and a server.
 - d) Unspecified Bit Rate (UBR) – a best effort service, whereby no amount of capacity is guaranteed and any cells may be discarded. UBR does not guarantee any throughput and uses only available bandwidth. This is referred to as “Class D” quality of service and is directed at delay-tolerant applications, such as file transfer and electronic mail.

2.4.6.4. ATM Network

An ATM network [16, 17, 19] consists of ATM switches and ATM endpoints (workstations, routers, etc) interconnected by point-to-point ATM links or interfaces, as shown by *figure 2.19*. ATM switches are responsible for cell transit through an ATM network, by accepting the incoming cell from an ATM endpoint or another ATM switch, reading and updating the cell header information and quickly switching the cell to an output interface towards its destination. To execute the transfer of cells, an association must be established between the two identities of the incoming and outgoing channels. This is performed with the aid of tables. These switching tables also define a logical connection through the network, while signaling is performed using ATM cells. The switching tables control switching through the nodes. ATM switches also have buffers, thus allowing the queuing of cells that cannot be immediately sent to the output, in case of two inputs trying to send cells to one and the same output simultaneously. ATM switches support two primary interfaces as follow [16, 17, 19]:

- a) User to network interface (UNI) – connects ATM end systems to an ATM switch.
- b) Network to network interface (NNI) – connects two ATM switches.

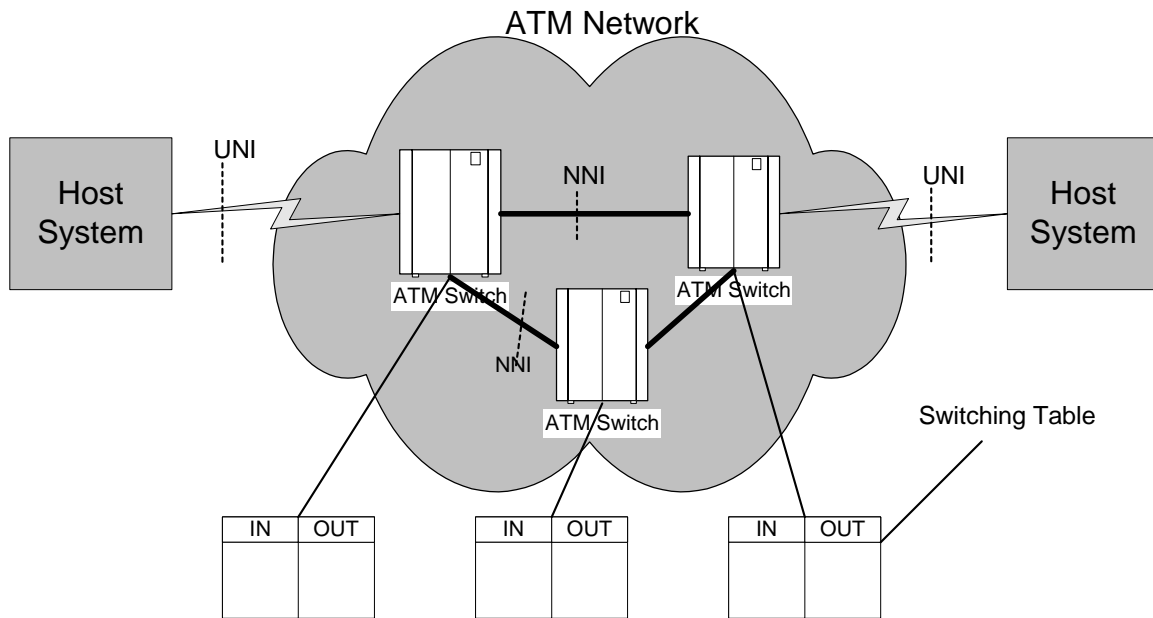


Figure 2.19, adapted from [16, 17, 19], shows the ATM Network

ATM 's strength as a transfer mode lies in its hardware-based extremely fast broadband switching that is superior to current router techniques and ordinary circuit switching techniques. ATM offers speeds of 155.52Mbps, 622.02Mbps, 2Gbps, and upwards.

2.4.6.5. ATM Protocol Structure

As compared to OSI-RM, ATM has eight (8) layers because the data link layer of OSI is divided in to two (2) layers in ATM namely ATM adaptation layer (AAL) and ATM layer [16, 17, 19], as shown by *figure 2.20*.

- a) ATM adaptation layer (AAL) – formats data and splits it into 48-byte chunks or cells – the process called segmentation and reassembly (SAR) – and sends them to the ATM layer.
- b) ATM layer – completes data formatting by addressing 5-byte header cells. Then the entire 53-byte cell is sent down to the physical layer.

ATM Reference Model

Application
Presentation
Session
Transport
Network
ATM Adaptation Layer (AAL)
ATM Layer
Physical

OSI-RM

Application
Presantation
Session
Transport
Network
Data Link
Physical

Figure 2.20, Adapted from [17, 19], shows ATM protocol layered structure compared to OSI-RM

2.4.6.6. Advantages and disadvantages of ATM

Advantages [16, 17, 19]:

- ATM is media-independent, thus operates on UTP, coax, STP and fiber optical cables
- Integrates different traffic types - voice, data and video - into one network hence improving efficiency and manageability
- Flexible bandwidth allocation - bandwidth on demand
- Simple routing and ensures reliability due to its connection-oriented nature
- Enables new applications due to its high speed and the integration of different traffic types
- Very scalable and flexible in geographic distances (from local to public and private wide area services), number of users, and access bandwidths.
- Cell switching is efficient and fast. The switch does not need to make allowances for variable lengths. It can easily clock the flow of cells.
- With fixed-size cells, traffic flow is predictable. You can accurately time the flow of cells because there are no variable-length cells that would throw such a calculation off.
- Because traffic is predictable, it is possible to guarantee that time-sensitive information will arrive on time, given that the network has enough capacity to carry it.

- j) ATM includes *QoS (quality of service)* features that let customers prioritize certain types of traffic, such as voice and video that must arrive on time, to ensure that less important traffic does not preempt real-time traffic.

Disadvantages [16, 17, 19]:

- a) One of the major drawbacks of ATM is the cost – it is very expensive
- b) Introduces overhead of cell header information – 5 bytes per cell; thus reducing bandwidth for the payload
- c) Possibility of cell discarding if switch is congested

2.5. Signaling

2.5.1. Introduction

As has been discussed in section 2.4, switching does not only concern the payload (user traffic), it also concerns the information used for management and control purposes. This switching of management and control information is called “signaling”; and has been defined in [19] as *“the exchange of information concerned with the establishment and control of connections and with management in a telecommunications network”*. Signaling systems are classified as [17, 19]:

- a) **In-Band Signaling** – involves sending the signaling information through the same communications channel as used for the information signal.
- b) **Out-Band Signaling** – involves sending the signaling information through the same communications channel as used for the information signal but using a different frequency range.

Exchanging control information over the same channel with the information signals was a key disadvantage and led to the following problems [19]: inefficiency of the utilization of the plant; slowness of signaling; and inability to signal in mid-call. The improvement was to provide separate signaling channels, which were however dedicated to specific user channels, the so called *“Channel-associated signaling (CAS)”* [19]. The improvement from CAS was the introduction of the *“Common Channel Signaling”* [19], which is a dedicated data network that acts as an overlay on the PSTN network and used solely for signaling. The PSTN gains access to the common channel signaling network at the signaling points. Two key signaling protocols used in telecommunications today are the signaling systems number 7 (SS7) and session initiation protocol (SIP); which are briefly discussed next.

2.5.2. Signaling System Number 7

Signaling System No.7 (SS7) developed by ITU-T [17, 19, 26], is a fully dedicated, self-contained, private, secure, high-speed, and highly reliable common channel signaling data network that is responsible for call setup, management, maintenance, billing, and call release in the PSTN. It is viewed as the nervous system of the telecommunications network [19]. It also links PSTN to IP networks and mobile networks. SS7 network utilizes two modes of operation, connection-oriented and connectionless, to provide other enhanced services such as local number portability (LNP), toll-free number calling, wireless services, mobile services, and many other features that support the IP networks as well as PSTN networks. Network elements in the public-switched telephone network use SS7 to exchange information used not only to set up calls but to affect routing and control the network. SS7 is a message-based system that operates on a separate digital line from the actual phone calls [17].

2.5.2.1. Signaling System No. 7 Architecture

SS7 network acts as an overlay to the PSTN network and comprises of switching units, databases (together called signaling points), routing nodes, and high-speed communication links; as follows [17, 19, 26]:

- a) **Signaling Points (SP)** – these are switches equipped with SS7 software that generate and terminate signaling messages. SPs communicate with other SPs and SCPs. Communication with other SPs occurs by means of exchanging messages to setup, manage, and release connections. SPs may also use transaction-related messages to contact databases in order to obtain information of how to route toll-free calls.
- b) **Signaling Control point (SCP)** – a network element or computer that contains necessary logic to service advanced call routing (toll-free number calling). An SCP may itself be a database computer or an interface connected to the database.
- c) **Signaling Transfer Point (STP)** – these are packet switches or routing nodes within the SS7 network that route network traffic between signaling points (SPs).

As with all other networks (PSTN or IP), all signaling points are identified by the point codes (PC) (as opposed to E.164 address in PSTN and IP addresses in IP networks). For routing to occur in SS7, each message must contain the originating PC (OPC) and the destination PC (DPC). Once this information is identified in each message, STPs can then route messages.

Traffic between signaling points is exchanged through high speed (56 or 64Kbps), dedicated, and bi-directional links. For availability and reliability, links between SPs are normally in pairs. SS7 signaling links are grouped according to their functions as follows [19, 26]:

- a) Access (A) Links – connect an SP (SSP, SCP) to an STP.
- b) Bridge (B) Links – connect two STPs together.
- c) Cross (C) Links – connect two STPs that perform the same functions into a mated pair. One STP is primary, while the other is backup, this is for reliability.
- d) Diagonal (D) Links – connect together STPs that are at different levels of hierarchy (i.e. a local STP to a secondary STP).
- e) Extended (E) Links – connect an SP to an alternative STP pair for reliability in case of failure of the primary pair.
- f) Fully (F) Links – connect two SPs together thus bypassing the STP.

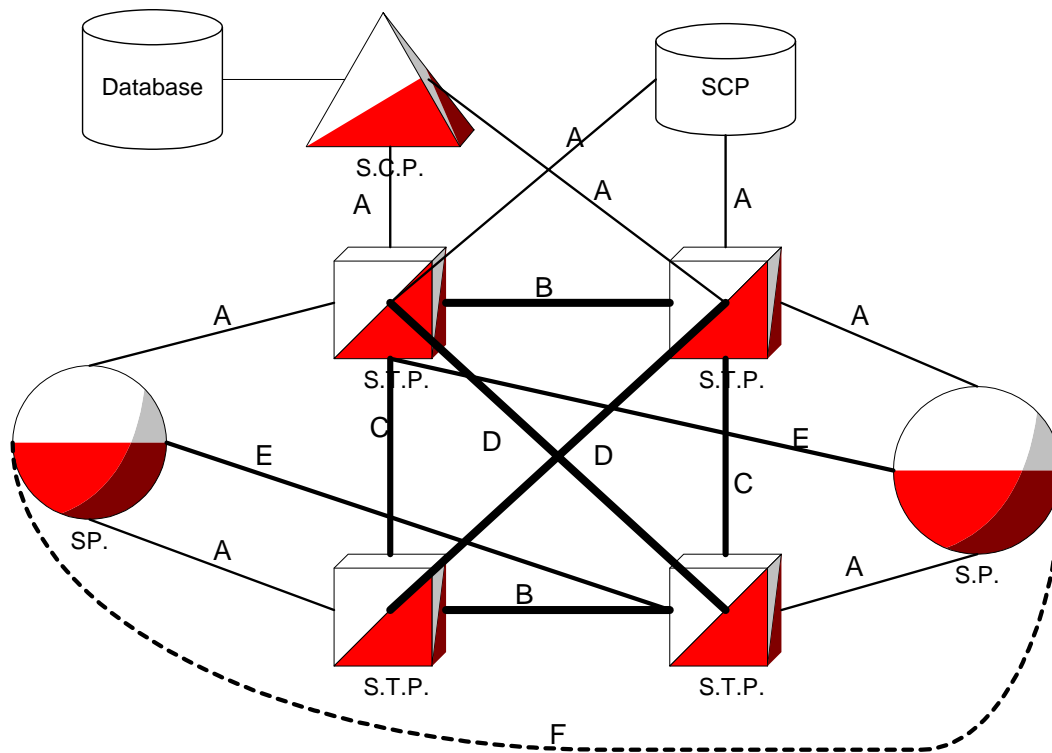


Figure 2.21, adapted from [19, 26], shows Signaling System Number 7 (SS7) Network Architecture

2.5.2.2. Signaling System No. 7 Protocol Structure

SS7 is layered protocol architecture [19, 26], with only four (4) layers when compared to OSI-RM. SS7 layers are grouped into message transfer part and user part. Message Transfer Part is concerned with the reliable transfer and routing of messages from higher layers across the SS7 network. The message transfer part comprises of three levels, MTP-1, MTP-2, and MTP-3 providing the physical, data link, and network layers respectively. On top of MTP-3, Signaling Connection and Control Part (SCCP) [19, 26] was introduced to provide connectionless and connection-oriented services and act as the transport layer to TCAP. SCCP also provides Global Title Translation (GTT). Together with MTP-3, they form the network layer. ISDN User Part (ISUP), falling under the user part, is the key signaling protocol of the SS7 network, and is used for call setup, control, and tear down (release). Transaction Capabilities Applications Part (TCAP), also falling under user part, supports transaction-oriented data transfer across the SS7 network. Such transactions include queries from SPs to SCPs and replies from SCPs to SPs. Applications supported by TCAP include Mobile Application Part (MAP), and Intelligent Network Application Part (INAP).

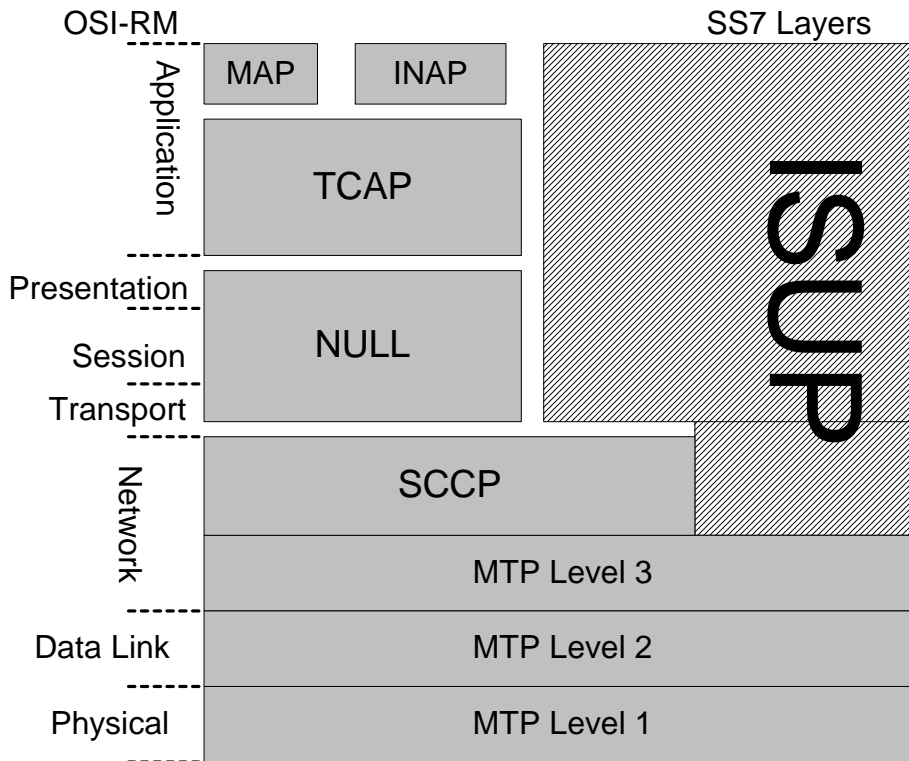


Figure 2.22, [19, 26], shows signaling system number 7 (SS7) protocol stack

2.5.2.3. Signaling System No.7 Message Types

Messages that traverse the SS7 network are grouped into signaling units (SUs) as follows [19, 26]:

- Fill-In Signaling Units (FISU)** – they are continuously being transmitted across the SS7 links when there are no other signaling units on the links. Their main aim is to keep the link alive, and they check link quality.
- Link Status Signaling Units (LSSU)** – used to give the status of the link between two SPs, by exchanging of link status messages between SPs.
- Message Signaling Units (MSU)** – these carry the actual call setup, control, and transaction-related messages across the SS7 network. ISUP and TCAP messages are carried within the MSUs.

2.5.3. Session Initiation Protocol

SIP, developed by IETF in 1996 [26, 27], is a text-based, highly extensible, web-dependent signaling and session management protocol. It got its name due the fact that its key purpose is to initiate, control and tear down sessions between users across packet-switched networks. It is said to be web-dependent as it is based on two of the web's protocols namely HTTP and SMTP. SIP represents telephone numbers as URLs [26, 27] (example SIP: l.kolobe@nui.ls), where

l.kolobe is the user at “nul.ls” domain. The domain can be either an IP address or the actual E.164 telephone number. Clicking on the SIP link or URL initiates a telephone call just like mailto in e-mails. Sessions in SIP can be either point-to-point or multicast and can carry all data types (data, video, voice, etc).

2.5.3.1. Session Initiation Protocol Architecture

A SIP network consists of the following elements [26, 27]:

- a) **User Agents (UA)** – these are the user end terminals that either initiate (hence called User Agent Clients) or terminate (hence called User Agent Servers) calls. Every UA is capable of acting as either a client or server but not as both in a single call.
- b) **Proxy Server** – these are intermediary devices that forward SIP messages on behalf of UAs to other elements in the SIP network.
- c) **Redirect Server** – simply tells a UAC where the UAS’s new location is. If a UAC cannot locate the UAS, it sends a request to the redirect server, which in turn responds with the new location of the UAS if known.
- d) **Registrar** – this is the server whereby SIP UAs register their current locations together with their addresses. The registrar then automatically tracks of UAs and keeps an updated database of locations and IP addresses.

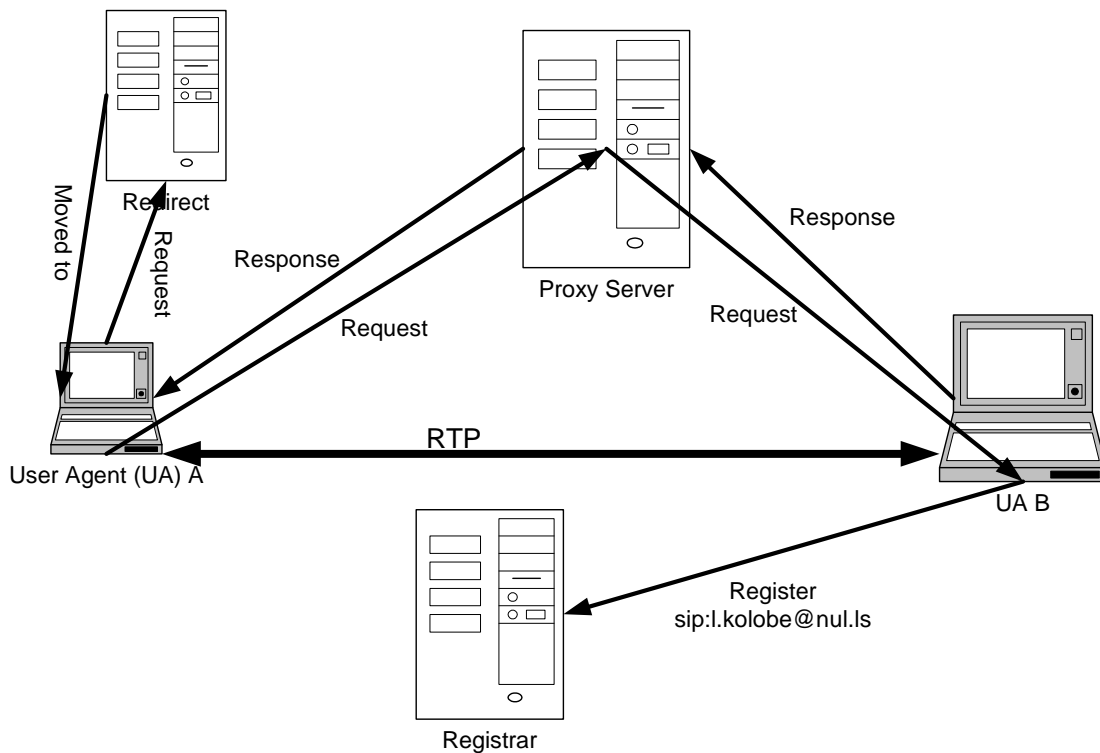


Figure 2.23 shows SIP architecture

2.5.3.2. Session Initiation Protocol Messages

SIP messages are grouped as either requests or responses. Examples of SIP requests are as follows [26, 27]:

- a) Invite – used by a UAC to request initiation of a session with the UAS.
- b) ACK – an acknowledgement or final confirmation that a session has been established.
- c) BYE – a request by either party to terminate the session.
- d) Option – used to query the called party about its capabilities.
- e) Register – sent by the UA to redirect server to inform it about its location.
- f) Cancel – an indication that a pending request must be cancelled.

Assignment:

Compare and contrast between two key signaling protocols discussed under section 2.5, namely SS7 and SIP. Your comparisons should be in the form of a technical report which must cover:

- a) Equipment used, End systems and the physical path
- b) Control processes
- c) Resource utilization
- d) Reliability, efficiency and security

Then based on the above comparisons, identify the trends in the telecommunications market and conclude, which protocol is best suited for the future.

Submission:

The hard-copy and the softcopy, no exceeding five typed pages, should be submitted on or before Friday November 9, 2007 at 11:50AM.